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ACOUSTICAL SOCIETY OF AMERICA

178th Meeting
of the Acoustical Society of America

Vol. 146 • No. 4 • Pt. 2 of 2 • 10.2019

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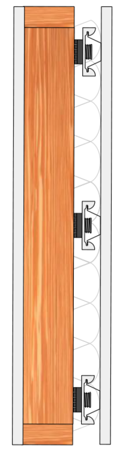

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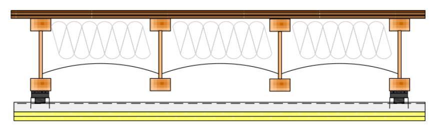

STC: 61
IIC: 58

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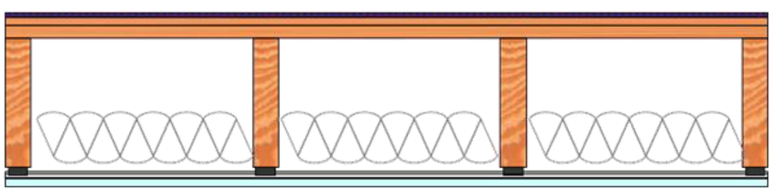

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CONTENTS

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TECHNICAL PROGRAM SUMMARY
178th Meeting of the Acoustical Society of America
2–6 December 2019
***Indicates Special Session**

MONDAY MORNING

*1aAA	Sustainable Acoustics for Smart Cities
1aAB	General Topics in Animal Bioacoustics I
*1aAO	Observational Acoustical Oceanography: A Look at Enabling Technology from Academia and Industry I
*1aBAa	Ultrasound Modeling Workshop - HITU Simulator
*1aBAb	Cavitation Nuclei: Bubbles, Droplets, and More I
*1aNS	Quiet Supersonic Flights 2018 I
1aPA	General Topics In Physical Acoustics I
*1aSA	Acoustics of 3D-Printed Materials and Structures
*1aSC	Universal and Experiential Influences on Phonetic Perception
*1aSP	Signal Processing for Architectural Acoustics and Noise Control I
1aUW	Underwater Acoustic Data Communication

MONDAY AFTERNOON

*1pAA	Sound Transmission and Impact Noise in Buildings
1pAB	General Topics in Animal Bioacoustics II
*1pAO	Observational Acoustical Oceanography: A Look at Enabling Technology from Academia and Industry II
*1pBA	Cavitation Nuclei: Bubbles, Droplets, and More II
*1pMUa	Asian Musical Instruments
*1pMUb	Concert: Musical Traditions of Thailand
*1pNS	Quiet Supersonic Flights 2018 II
1pPA	General Topics in Physical Acoustics II
*1pSA	Acoustic Metamaterials
*1pSC	Self-Perception in Speech Production
*1pSP	Signal Processing for Architectural Acoustics and Noise Control II
1pUWa	Scattering and Reflection
1pUWb	Detection, Classification and Noise

MONDAY EVENING

*1eID	Tutorial Lecture on Effective Media Interactions Training Workshop

TUESDAY MORNING

*2aAA	Computational Acoustics for Architectural Applications
*2aAB	Low-Frequency Sound Production and Passive Acoustic Monitoring I
*2aAOa	Session in Honor of Michael Buckingham I
*2aAOb	Acoustical Oceanography Prize Lecture
*2aBAa	Application of Quantitative Ultrasound in vivo in Humans I
*2aBAb	High Frame Rate Ultrasound Imaging: Technical Developments and Clinical Applications I
*2aEA	Acoustical Engineering in Consumer Electronics
*2aEDa	Mentoring Graduate and Undergraduate Students
*2aEDb	Acoustics Education Prize Lecture
*2aMU	Experimental Methods in Musical Acoustics: Best Practices
*2aNSa	Effects on People and Wildlife from Transportation Noise (Land, Air, Sea), as Well as Innovative Solutions for Reducing Noise
*2aNSb	Outdoor Entertainment Noise
*2aPA	Design of Acoustics Metamaterials: Optimization and Machine Learning I
2aPP	Current Topics in Physiological and Psychological Acoustics (Poster Session)
*2aSA	Flow-Induced Vibration and Noise
2aSC	Second-Language Acquisition and Bilingualism (Poster Session)
2aSP	General Topics in Signal Processing II
*2aUW	Comprehensive Nuclear-Test-Ban Treaty International Monitoring System Global Sensor Network: Scientific Aspects and Civil Applications

TUESDAY AFTERNOON

*2pAA	Large Music Rehearsing Spaces
*2pAB	Low-Frequency Sound Production and Passive Acoustic Monitoring II
*2pAO	Session in Honor of Michael Buckingham II
*2pBAa	High Frame Rate Ultrasound Imaging: Technical Developments and Clinical Applications II
*2pBAb	Application of Quantitative Ultrasound in vivo in Humans II
2pEA	Engineering Acoustics Mixtape: Microfluidics, Arrays, Pipes, Ducts, & Damping
*2pIDa	Guidance From the Experts: Applying for Grants and Fellowships
*2pIDb	Introduction to Technical Committees
*2pNSa	Current Trends and Advancements in Applying Acoustics to Smart Cities
2pNSb	General Noise Session
*2pPA	Design of Acoustics Metamaterials: Optimization and Machine Learning II
*2pPPa	Open Source Audio Processing Tools for Hearing Research I
*2pPPb	Open Source Audio Processing Tools for Hearing Research II (Demonstrations)
2pSA	Characterization and Analysis of System Properties
*2pSP	Signal Processing for Biological Transients
2pUW	Propagation and Sound Generation Physics and Modeling

WEDNESDAY MORNING

*3aAA	Assembly Space Renovation Challenges I
*3aAB	Urban Noise: Its Effects on Animals' Acoustic Communication
*3aAO	Bioacoustics and Acoustical Oceanography: 20 Years Later I
3aBA	New Frontiers in Doppler Ultrasound
*3aCA	Application of Model Reduction in Computational Acoustics
*3aED	Hands-On Demonstrations for Middle- and High-School Students
3aMU	General Topics in Musical Acoustics I
3aNS	Community Noise
*3aPA	Non-Reciprocal and Topological Acoustics
3aPP	Technological Advancements in Hearing Research (Poster Session)
*3aSA	Novel Methods For Energy Dissipation in Structures
3aSC	Speech and Hearing Disorders and Child Speech (Poster Session)
*3aSP	Memorial Session in Honor of Ed Sullivan
3aUWa	Sound Interaction with the Seabed
*3aUWb	A.B. Wood Medal and Prize Lecture

WEDNESDAY AFTERNOON

*3pAA	Assembly Space Renovation Challenges II
*3pABa	Standards in Animal Bioacoustics - Purpose, Need, and Application
3pABb	Animal Bioacoustics Poster Session
*3pAO	Bioacoustics and Acoustical Oceanography: 20 Years Later II
3pBA	Biomedical Acoustics Poster Session
*3pEA	Acoustic Holography and Visualization of Sound: Methods and Applications
*3pID	Hot Topics in Acoustics
*3pMU	Machine Learning in Musical Acoustics
*3pNS	Development of New Sounds for Electric Vehicles
3pPA	General Topics in Physical Acoustics III
3pSA	Topics on Health Monitoring and Non-Destructive Testing
3pSC	Neuroscience of Speech Production and Perception and Speech Technology (Poster Session)
3pSP	General Topics in Signal Processing (Poster Session)
3pUWa	Source and System Component Localization
3pUWb	Underwater Acoustics Topics (Poster Session)

WEDNESDAY EVENING

*3eED	Listen-Up and Get Involved!

THURSDAY MORNING

*4aAA	Architectural Soundscapes I
4aAB	Applications of Machine Learning to Bioacoustics I
4aAO	Acoustic Propagation and Geoacoustic Inversion
*4aBAa	Cavitation Bioeffects I
4aBAb	Biomedical Acoustics I

4aEA Transducers
4aMU General Topics in Musical Acoustics II
*4aNS Supersonic Jet Aeroacoustics I
*4aPA Aqueous Acoustic Metamaterials I
*4aSA Computational Methods for Mid-Frequency Structural Acoustic Problems
4aSC Speech Production (Poster Session)
*4aSPa Eco-Active Sonar
4aSPb Array Signal Processing I
4aUW Measurements, Instrumentation and Analysis Methods

THURSDAY AFTERNOON

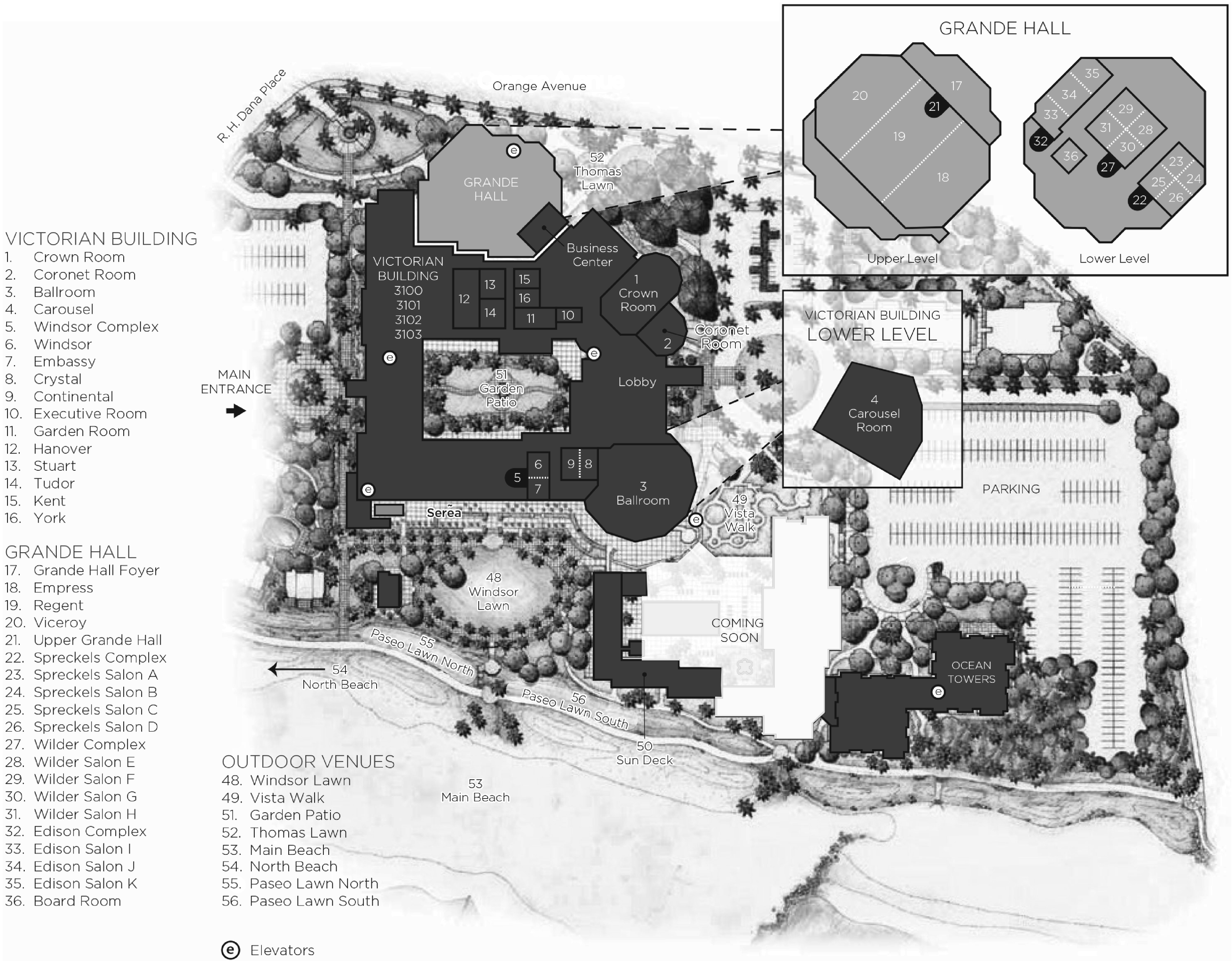
*4pAA Architectural Soundscapes II
*4pAB Applications of Machine Learning to Bioacoustics II
4pAOa Underwater Noise
4pAOb Underwater Scattering
4pBAa Biomedical Acoustics II
*4pBAb Cavitation Bioeffects II
*4pCA Parabolic Equation Methods Across Acoustics
*4pEDa Selecting a Textbook for Teaching an Acoustics Course
*4pEDb Take 5's
*4pNS Supersonic Jet Aeroacoustics II
*4pPA Aqueous Acoustic Metamaterials II
4pPPa Perceptual Processing of Sound
4pPPb Speech and Pitch Perception
4pSC Speech Perception (Poster Session)
4pSP Array Signal Processing II
*4pUW Ship Source Level Estimation: Methods and Measurements

FRIDAY MORNING

*5aAA How Does Speech Perception Work: A Tutorial and Panel Discussion for Architectural Speech Privacy
*5aAO Marine Seismoacoustics
*5aBA Ultrasound Phantom Development and Tissue Characterization
5aCA General Topics in Computational Acoustics
*5aNS Jet Noise Reduction Workshop
5aPA General Topics in Physical Acoustics IV
5aSA General Topics in Structural Acoustics and Vibration
5aSC Speech Articulation (Poster Session)
5aUW Remote Sensing, Inversion, and Passive Sensing

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/PM
Carousel				2aEDa 7:45 2aEDb 11:00	2pIDa 1:00 2pIDb 2:40		3aED 8:30		3eED 5:30		4pEDa 1:00 4pEDb 3:00		
Continental				2aNSb 8:30									
Crystal				2aNSa 8:30									
Continental/ Crystal	1aNS 9:15	1pNS 1:15			2pNSa 1:00 2pNSb 3:00		3aNS 7:55	3pNS 1:00		4aNS 9:00	4pNS 1:00	TCNS 7:30	5aNS 9:00–4.15
Coronet	1aBAa 8:15	1pMUa 1:45 1pMUb 4:30		2aMU 8:00	2pPPa 1:15		3aMU 9:00	3pMU 1:30		4aMU 9:00	4pPPa 1:00 4pPPb 3:00	TCMU 7:30	
Crown	1aSC 9:00	1pSC 1:15		2aPP 9:00 2aSC 9:00	2pPPb 3:15		3aPP 9:00 3aSC 9:00	3pABb 3pBA 3pSC 3pSP 3pUWb 1:00		4aSC 8:00	4pSC 1:15		5aSC 8:00
Edison	1aAB 9:00	1pAB 1:30		2aAB 8:00	2pAB 1:30	TCAB 7:30	3aAB 8:00	3pABa 1:00		4aAB 8:00	4pAB 1:30		5aBA 7:30
Empress	1aAO 8:15	1pAO 1:15		2aAOa 8:30 2aAOb 11:00	2pAO 1:15	TCAO 7:30	3aAO 8:00	3pAO 1:00		4aAO 8:15	4pAOa 1:15 4pAOb 3:15		5aAO 8:30
Garden				2aBAa 8:00	2pBAb 1:15	TCPP 7:30	3aBA 7:55			4aBAa 8:00	4pBAb 1:30		
Hanover	1aBAb 10:00	1pBA 1:15		2aBAb 9:00	2pBAa 1:15			3pID 1:00	TCBA 7:30	4aBAb 10:30	4pBAa 1:15		5aCA 8:30
Regent	1aSP 9:45	1pSP 1:00	1eID 7:00	2aSP 8:00	2pSP 1:00	TCSP 7:30	3aSP 8:00			4aSPa 7:55 4aSPb 9:55	4pSP 1:15	TCSC 7:30	
Spreckels	1aSA 8:45	1pSA 1:00		2aSA 8:45	2pSA 1:15	TCSA 7:30	3aSA 8:30	3pSA 1:30		4aSA 8:00			5aSA 8:00
Stuart				2aEA 7:45	2pEA 1:15		3aCA 8:00	3pEA 1:00		4aEA 8:00	4pCA 1:15		
Viceroy	1aUW 9:00	1pUWa 1:00 1pUWb 3:15		2aUW 8:00	2pUW 1:00		3aUWa 8:00 3aUWb 11:00	3pUWa 1:00		4aUW 8:15	4pUW 1:00	TCUW 7:30	5aUW 8:15
Wilder	1aPA 9:00	1pPA 1:15		2aPA 7:45	2pPA 1:30	TCPA 7:30	3aPA 9:15	3pPA 1:00		4aPA 9:00	4pPA 1:15		5aPA 8:00
Windsor	1aAA 9:45	1pAA 1:00		2aAA 8:00	2pAA 1:00	TCAA 7:30	3aAA 8:00	3pAA 1:00		4aAA 8:30	4pAA 1:45		5aAA 9:00


GRANDE HALL
20
17
21
19
18
Upper Level
35
34
33
29
31
28
30
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36
27
23
25
24
26
22
Lower Level
VICTORIAN BUILDING
LOWER LEVEL
4
Carousel
Room
Orange Avenue
R. H. Dana Place
GRANDE
HALL
52
Thomas
Lawn
Business
Center
VICTORIAN
BUILDING
3100
3101
3102
3103
12
13
14
15
16
11
10
1
Crown
Room
2
Coronet
Room
Lobby
51
Garden
Patio
MAIN
ENTRANCE
5
6
7
9
8
3
Ballroom
Serea
49
Vista
Walk
PARKING
48
Windsor
Lawn
COMING
SOON
OCEAN
TOWERS
55
Paseo Lawn North
54
North Beach
56
Paseo Lawn South
50
Sun Deck
53
Main Beach
VICTORIAN BUILDING
1. Crown Room
2. Coronet Room
3. Ballroom
4. Carousel
5. Windsor Complex
6. Windsor
7. Embassy
8. Crystal
9. Continental
10. Executive Room
11. Garden Room
12. Hanover
13. Stuart
14. Tudor
15. Kent
16. York
GRANDE HALL
17. Grande Hall Foyer
18. Empress
19. Regent
20. Viceroy
21. Upper Grande Hall
22. Spreckels Complex
23. Spreckels Salon A
24. Spreckels Salon B
25. Spreckels Salon C
26. Spreckels Salon D
27. Wilder Complex
28. Wilder Salon E
29. Wilder Salon F
30. Wilder Salon G
31. Wilder Salon H
32. Edison Complex
33. Edison Salon I
34. Edison Salon J
35. Edison Salon K
36. Board Room
OUTDOOR VENUES
48. Windsor Lawn
49. Vista Walk
51. Garden Patio
52. Thomas Lawn
53. Main Beach
54. North Beach
55. Paseo Lawn North
56. Paseo Lawn South
e Elevators

TECHNICAL PROGRAM CALENDAR
178th Meeting of the Acoustical Society of America
2–6 December 2019

MONDAY MORNING

9:45 1aAA **Architectural Acoustics and Noise:** Sustainable Acoustics for Smart Cities. Windsor

9:00 1aAB **Animal Bioacoustics:** General Topics in Animal Bioacoustics I. Edison

8:15 1aAO **Acoustical Oceanography and ASA Committee on Standards:** Observational Acoustical Oceanography: A Look at Enabling Technology from Academia and Industry I. Empress

8:15 1aBAa **Biomedical Acoustics:** Ultrasound Modeling Workshop - HITU Simulator. Coronet

10:00 1aBAb **Biomedical Acoustics and Physical Acoustics:** Cavitation Nuclei: Bubbles, Droplets, and More I. Hanover

9:15 1aNS **Noise and Physical Acoustics:** Quiet Supersonic Flights 2018 I. Continental/ Crystal

9:00 1aPA **Physical Acoustics:** General Topics in Physical Acoustics I. Wilder

8:45 1aSA **Structural Acoustics and Vibration:** Acoustics of 3D-Printed Materials and Structures. Spreckles

9:00 1aSC **Speech Communication:** Universal and Experiential Influences on Phonetic Perception. Crown

9:45 1aSP **Signal Processing in Acoustics, Architectural Acoustics, and Noise:** Signal Processing for Architectural Acoustics and Noise Control I. Regent

9:00 1aUW **Underwater Acoustics:** Underwater Acoustic Data Communication. Viceroy

MONDAY AFTERNOON

1:00 1pAA **Architectural Acoustics, Noise, Structural Acoustics and Vibration, and ASA Committee on Standards:** Sound Transmission and Impact Noise in Buildings. Windsor

1:30 1pAB **Animal Bioacoustics:** General Topics in Animal Bioacoustics II. Edison

1:15 1pAO **Acoustical Oceanography and ASA Committee on Standards:** Observational Acoustical Oceanography: A Look at Enabling Technology from Academia and Industry II. Empress

1:15 1pBA **Biomedical Acoustics and Physical Acoustics:** Cavitation Nuclei: Bubbles, Droplets, and More II. Hanover

1:45 1pMUa **Musical Acoustics:** Asian Musical Instruments. Coronet

4:30 1pMUb **Musical Acoustics:** Concert: Musical Traditions of Thailand. Coronet

1:15 1pNS **Noise and Physical Acoustics:** Quiet Supersonic Flights 2018 II. Continental/ Crystal

1:15 1pPA **Physical Acoustics:** General Topics in Physical Acoustics II. Wilder

1:00 1pSA **Structural Acoustics and Vibration, Physical Acoustics, and Signal Processing in Acoustics:** Acoustic Metamaterials. Spreckles

1:15 1pSC **Speech Communication:** Self-Perception in Speech Production. Crown

1:00 1pSP **Signal Processing in Acoustics, Architectural Acoustics, and Noise:** Signal Processing for Architectural Acoustics and Noise Control II. Regent

1:00 1pUWa **Underwater Acoustics:** Scattering and Reflection. Viceroy

3:15 1pUWb **Underwater Acoustics:** Detection, Classification, and Noise. Viceroy

MONDAY EVENING

7:00 1eID **Interdisciplinary:** Tutorial Lecture on Effective Media Interactions Training Workshop. Regent

TUESDAY MORNING

8:00 2aAA **Architectural Acoustics and Computational Acoustics:** Computational Acoustics for Architectural Applications. Windsor

8:00 2aAB **Animal Bioacoustics and Acoustical Oceanography:** Low-Frequency Sound Production and Passive Acoustic Monitoring I. Edison

8:30 2aAOa **Acoustical Oceanography and Underwater Acoustics:** Session in Honor of Michael Buckingham I. Empress

11:00 2aAOb **Acoustical Oceanography:** Acoustical Oceanography Prize Lecture. Empress

8:00 2aBAa **Biomedical Acoustics:** Application of Quantitative Ultrasound in vivo in Humans I. Garden

9:00 2aBAb **Biomedical Acoustics and Signal Processing in Acoustics:** High Frame Rate Ultrasound Imaging: Technical Developments and Clinical Applications I. Hanover

7:45 2aEA **Engineering Acoustics and Structural Acoustics and Vibration:** Acoustical Engineering in Consumer Electronics. Stuart

7:45 2aEDa **Education in Acoustics, Student Council, and Women in Acoustics:** Mentoring Graduate and Undergraduate Students. Carousel

11:00 2aEDb **Education in Acoustics:** Acoustics Education Prize Lecture. Carousel

8:00 2aMU **Musical Acoustics:** Experimental Methods in Musical Acoustics: Best Practices. Coronet

8:30 2aNSa **Noise, Animal Bioacoustics and Underwater Acoustics:** Effects on People and Wildlife from Transportation Noise (Land, Air, Sea), as Well as Innovative Solutions for Reducing Noise. Crystal

8:30 2aNSb **Noise:** Outdoor Entertainment Noise. Continental

7:45 2aPA **Physical Acoustics, Structural Acoustics and Vibration, and Computational Acoustics:** Design of Acoustics Metamaterials: Optimization and Machine Learning I. Wilder

9:00 2aPP **Psychological and Physiological Acoustics:** Current Topics in Physiological and Psychological Acoustics (Poster Session). Crown

8:45 2aSA **Structural Acoustics and Vibration and Noise:** Flow-Induced Vibration and Noise. Spreckles

9:00 2aSC **Speech Communication:** Second-Language Acquisition and Bilingualism (Poster Session). Crown

8:00 2aSP **Signal Processing in Acoustics:** General Topics in Signal Processing II. Regent

8:00 2aUW **Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics:** Comprehensive Nuclear-Test-Ban Treaty International Monitoring System Global Sensor Network: Scientific Aspects and Civil Applications. Viceroy

TUESDAY AFTERNOON

1:00 2pAA **Architectural Acoustics and Musical Acoustics:** Large Music Rehearsing Spaces. Windsor

1:30 2pAB **Animal Bioacoustics and Acoustical Oceanography:** Low-Frequency Sound Production and Passive Acoustic Monitoring II. Edison

1:15 2pAO **Acoustical Oceanography and Underwater Acoustics:** Session in Honor of Michael Buckingham II. Empress

1:15 2pBAa **Biomedical Acoustics and Signal Processing in Acoustics:** High Frame Rate Ultrasound Imaging: Technical Developments and Clinical Applications II. Hanover

1:15 2pBAb **Biomedical Acoustics:** Application of Quantitative Ultrasound in vivo in Humans II. Garden

1:15 2pEA **Engineering Acoustics:** Engineering Acoustics Mixtape: Microfluidics, Arrays, Pipes, Ducts, and Damping. Stuart

1:00 2pIDa **Interdisciplinary and Student Council:** Guidance From the Experts: Applying for Grants and Fellowships. Carousel

2:40 2pIDb **Interdisciplinary:** Introduction to Technical Committees. Carousel

1:00 2pNSa **Noise, Architectural Acoustics, and ASA Committee on Standards:** Current Trends and Advancements in Applying Acoustics to Smart Cities. Continental/Crystal

3:00 2pNSb **Noise:** General Noise Session. Continental/Crystal

1:30 2pPA **Physical Acoustics, Structural Acoustics and Vibration, and Computational Acoustics:** Design of Acoustics Metamaterials: Optimization and Machine Learning II. Wilder

1:15 2pPPa **Psychological and Physiological Acoustics and Speech Communication:** Open Source Audio Processing Tools for Hearing Research I. Coronet

3:15 2pPPb **Psychological and Physiological Acoustics and Speech Communication:** Open Source Audio Processing Tools for Hearing Research II (Demonstrations). Crown

1:15 2pSA **Structural Acoustics and Vibration:** Characterization and Analysis of System Properties. Spreckles

1:00 2pSP **Signal Processing in Acoustics, Animal Bioacoustics, Underwater Acoustics, Acoustical Oceanography, and Speech Communication:** Signal Processing for Biological Transients. Regent

1:00 2pUW **Underwater Acoustics:** Propagation and Sound Generation Physics and Modeling. Viceroy

WEDNESDAY MORNING

8:00 3aAA **Architectural Acoustics:** Assembly Space Renovation Challenges I. Windsor

8:00 3aAB **Animal Bioacoustics:** Urban Noise: Its Effects on Animals' Acoustic Communication. Edison

8:00 3aAO **Acoustical Oceanography and Animal Bioacoustics:** Bioacoustics and Acoustical Oceanography: 20 Years Later I. Empress

7:55 3aBA **Biomedical Acoustics:** New Frontiers in Doppler Ultrasound. Garden

8:00 3aCA **Computational Acoustics, Structural Acoustics and Vibration, and Signal Processing in Acoustics:** Application of Model Reduction in Computational Acoustics. Stuart

8:30 3aED **Education in Acoustics:** Hands-On Demonstrations for Middle- and High-School Students. Carousel

9:00 3aMU **Musical Acoustics:** General Topics in Musical Acoustics I. Coronet

7:55 3aNS **Noise and ASA Committee on Standards:** Community Noise. Continental/Crystal

9:15 3aPA **Physical Acoustics and Engineering Acoustics:** Non-Reciprocal and Topological Acoustics. Wilder

9:00 3aPP **Psychological and Physiological Acoustics:** Technological Advancements in Hearing Research (Poster Session). Crown

8:30 3aSA **Structural Acoustics and Vibration, Noise, Physical Acoustics, and Engineering Acoustics:** Novel Methods For Energy Dissipation in Structures. Spreckles

9:00 3aSC **Speech Communication:** Speech and Hearing Disorders and Child Speech (Poster Session). Crown

8:00 3aSP **Signal Processing in Acoustics and Underwater Acoustics:** Memorial Session in Honor of Ed Sullivan. Regent

8:00 3aUWa **Underwater Acoustics:** Sound Interaction with the Seabed. Viceroy

11:00 3aUWb **Underwater Acoustics and Acoustical Oceanography:** A.B. Wood Medal and Prize Lecture. Viceroy

WEDNESDAY AFTERNOON

1:00 3pAA **Architectural Acoustics:** Assembly Space Renovation Challenges II. Windsor

1:00 3pABa **Animal Bioacoustics and ASA Committee on Standards:** Standards in Animal Bioacoustics — Purpose, Need, and Application. Edison

1:00 3pABb **Animal Bioacoustics:** Animal Bioacoustics Poster Session. Crown

1:00 3pAO **Acoustical Oceanography and Animal Bioacoustics:** Bioacoustics and Acoustical Oceanography: 20 years Later II. Empress

1:00 3pBA **Biomedical Acoustics:** Biomedical Acoustics Poster Session. Crown

1:00 3pEA **Engineering Acoustics, Physical Acoustics, and Structural Acoustics and Vibration:** Acoustic Holography and Visualization of Sound: Methods and Applications. Stuart

1:00 3pID **Interdisciplinary:** Hot Topics in Acoustics. Hanover

1:30 3pMU **Musical Acoustics, Signal Processing in Acoustics, and Computational Acoustics:** Machine Learning in Musical Acoustics. Coronet

1:00 3pNS **Noise, Psychological and Physiological Acoustics, and ASA Committee on Standards:** Development of New Sounds for Electric Vehicles. Continental/Crystal

1:00 3pPA **Physical Acoustics:** General Topics in Physical Acoustics III. Wilder

1:30 3pSA **Structural Acoustics and Vibration:** Topics on Health Monitoring and Non-Destructive Testing. Spreckles

1:00 3pSC **Speech Communication:** Neuroscience of Speech Production and Perception and Speech Technology (Poster Session). Crown

1:00 3pSP **Signal Processing in Acoustics:** General Topics in Signal Processing (Poster Session). Crown

1:00 3pUWa **Underwater Acoustics:** Source and System Component Localization. Viceroy

1:00 3pUWb **Underwater Acoustics:** Underwater Acoustics Topics (Poster Session). Crown

WEDNESDAY EVENING

5:30 3eED **Education in Acoustics and Women in Acoustics:** Listen-Up and Get Involved! Carousel

THURSDAY MORNING

8:30 4aAA **Architectural Acoustics and Noise:** Architectural Soundscapes I. Windsor

8:00 4aAB **Animal Bioacoustics, Signal Processing in Acoustics, Acoustical Oceanography, and Computational Acoustics:** Applications of Machine Learning to Bioacoustics I. Edison

8:15 4aAO **Acoustical Oceanography:** Acoustic Propagation and Geoacoustic Inversion. Empress

8:00 4aBAa **Biomedical Acoustics:** Cavitation Bioeffects I. Garden

10:30 4aBAb **Biomedical Acoustics:** Biomedical Acoustics I. Hanover

8:00 4aEA **Engineering Acoustics:** Transducers. Stuart

9:00 4aMU **Musical Acoustics:** General Topics in Musical Acoustics II. Coronet

9:00 4aNS **Noise, Physical Acoustics, and Computational Acoustics:** Supersonic Jet Aeroacoustics I. Continental/Crystal

9:00 4aPA **Physical Acoustics, Engineering Acoustics, Structural Acoustics and Vibration, and Underwater Acoustics:** Aqueous Acoustic Metamaterials I. Wilder

8:00 4aSA **Structural Acoustics and Vibration and Computational Acoustics:** Computational Methods for Mid-Frequency Structural Acoustic Problems. Spreckles

8:00 4aSC **Speech Communication:** Speech Production (Poster Session). Crown

7:55 4aSPa **Signal Processing in Acoustics and Animal Bioacoustics:** Eco-Active Sonar. Regent

9:55 4aSPb **Signal Processing in Acoustics:** Array Signal Processing I. Regent

8:15 4aUW **Underwater Acoustics:** Measurements, Instrumentation and Analysis Methods. Viceroy

THURSDAY AFTERNOON

1:45 4pAA **Architectural Acoustics and Noise:** Architectural Soundscapes II. Windsor

1:30 4pAB **Animal Bioacoustics, Signal Processing in Acoustics, Acoustical Oceanography, and Computational Acoustics:** Applications of Machine Learning to Bioacoustics II. Edison

1:15 4pAOa **Acoustical Oceanography:** Underwater Noise. Empress

3:15 4pAOb **Acoustical Oceanography:** Underwater Scattering. Empress

1:15 4pBAa **Biomedical Acoustics:** Biomedical Acoustics II. Hanover

1:30 4pBAb **Biomedical Acoustics:** Cavitation Bioeffects II. Garden

1:15 4pCA **Computational Acoustics, Physical Acoustics, Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Structural Acoustics and Vibration:** Parabolic Equation Methods Across Acoustics. Stuart

1:00 4pEDa **Education in Acoustics and Musical Acoustics:** Selecting a Textbook for Teaching an Acoustics Course. Carousel

3:00 4pEDb **Education in Acoustics:** Take 5’s. Carousel

1:00 4pNS **Noise, Physical Acoustics, and Computational Acoustics:** Supersonic Jet Aeroacoustics II. Continental/Crystal

1:15 4pPA **Physical Acoustics, Engineering Acoustics, Structural Acoustics and Vibration, and Underwater Acoustics:** Aqueous Acoustic Metamaterials II. Wilder

1:00 4pPPa **Psychological and Physiological Acoustics and Speech Communication:** Perceptual Processing of Sound. Coronet

3:00 4pPPb **Psychological and Physiological Acoustics:** Speech and Pitch Perception. Coronet

1:15 4pSC **Speech Communication:** Speech Perception (Poster Session). Crown

1:15 4pSP **Signal Processing in Acoustics:** Array Signal Processing II. Regent

1:00 4pUW **Underwater Acoustics and Animal Bioacoustics:** Ship Source Level Estimation: Methods and Measurements. Viceroy

FRIDAY MORNING

9:00 5aAA **Architectural Acoustics, Psychological and Physiological Acoustics, Speech Communication, and ASA Committee on Standards:** How Does Speech Perception Work: A Tutorial and Panel Discussion for Architectural Speech Privacy. Windsor

8:30 5aAO **Acoustical Oceanography and Underwater Acoustics:** Marine Seismoacoustics. Empress

7:30 5aBA **Biomedical Acoustics:** Ultrasound Phantom Development and Tissue Characterization. Edison

8:30 5aCA **Computational Acoustics:** General Topics in Computational Acoustics. Hanover

9:00 5aNS **Noise, Physical Acoustics, and Computational Acoustics:** Jet Noise Reduction Workshop. Continental/Crystal

8:00 5aPA **Physical Acoustics:** General Topics in Physical Acoustics IV. Wilder

8:00 5aSA **Structural Acoustics and Vibration:** General Topics in Structural Acoustics and Vibration. Spreckles

8:00 5aSC **Speech Communication:** Speech Articulation (Poster Session). Crown

8:15 5aUW **Underwater Acoustics:** Remote Sensing, Inversion, and Passive Sensing. Viceroy

SCHEDULE OF COMMITTEE MEETINGS AND OTHER EVENTS

ASA COUNCIL AND ADMINISTRATIVE COMMITTEES

(O) = Meetings and events that will be held at outdoor locations

Mon, 2 December, 7:30 a.m.	Executive Council	Garden
Mon, 2 December, 3:30 p.m.	Technical Council	Garden
Tue, 3 December, 7:00 a.m.	ASA Books	Serea PDR
Tue, 3 December, 7:30 a.m.	Panel on Public Policy	Tudor
Tue, 3 December, 11:45 a.m.	Editorial Board	Vista Walk (O)
Tue, 3 December, 12:00 noon	Student Council	Serea PDR
Tue, 3 December, 12:30 p.m.	Prizes & Special Fellowships	Board
Tue, 3 December, 1:30 p.m.	Meetings	Kent/York
Tue, 3 December, 4:00 p.m.	Newman Fund Advisory	Executive
Tue, 3 December, 4:00 p.m.	International Liaison	Tudor
Tue, 3 December, 5:00 p.m.	Education in Acoustics	Hanover
Tue, 3 December, 5:00 p.m.	Women in Acoustics	Wilder
Wed, 4 December, 7:00 a.m.	Archives & History	Kent
Wed, 4 December, 7:00 a.m.	College of Fellows	Executive
Wed, 4 December, 7:00 a.m.	International Research & Education	York
Wed, 4 December, 7:00 a.m.	Publication Policy	Tudor
Wed, 4 December, 7:00 a.m.	Regional and Student Chapters	Vista Walk (O)
Wed, 4 December, 7:30 a.m.	Finance	Serea PDR
Wed, 4 December, 9:30 a.m.	AS Foundation Board	Executive
Wed, 4 December, 11:30 a.m.	Public Relations	Executive
Wed, 4 December, 12:00 noon	Medals and Awards	Serea PDR
Wed, 4 December, 12:00 noon	Membership	Kent/York
Wed, 4 December, 5:45 p.m.	TCAA Speech Privacy	Garden
Thu, 5 December, 7:00 a.m.	Member Engagement	Kent/York
Thu, 5 December, 7:30 a.m.	Investments	Serea PDR
Thu, 5 December, 7:30 a.m.	Tutorials, Short Courses, Hot Topics	Executive
Thu, 5 December, 9:30 a.m.	Financial Affairs Admin Council	Serea PDR
Thu, 5 December, 2:00 p.m.	Strategic Plan Champions	Ballroom
Thu, 5 December, 4:30 p.m.	Member Engagement and Diversity Admin Council	Executive
Thu, 5 December, 4:30 p.m.	Outreach Admin Council	Board
Thu, 5 December, 4:30 p.m.	Publications and Standards Admin Council	Kent/York
Fri, 6 December, 7:00 a.m.	Technical Council	Garden
Fri, 6 December, 11:00 a.m.	Executive Council	Garden

TECHNICAL COMMITTEE OPEN MEETINGS

Tue, 3 December, 4:45 p.m.	Engineering Acoustics	Stuart
Tue, 3 December, 7:30 p.m.	Acoustical Oceanography	Empress
Tue, 3 December, 7:30 p.m.	Animal Bioacoustics	Edison
Tue, 3 December, 7:30 p.m.	Architectural Acoustics	Windsor
Tue, 3 December, 7:30 p.m.	Physical Acoustics	Wilder
Tue, 3 December, 7:30 p.m.	Psychological and Physiological Acoustics	Garden
Tue, 3 December, 7:30 p.m.	Structural Acoustics and Vibration	Spreckels
Wed, 4 December, 7:30 p.m.	Biomedical Acoustics	Hanover
Wed, 4 December, 7:30 p.m.	Signal Processing in Acoustics	Empress
Thu, 5 December, 4:30 p.m.	Computational Acoustics	Stuart
Thu, 5 December, 7:30 p.m.	Musical Acoustics	Coronet
Thu, 5 December, 7:30 p.m.	Noise	Crystal/Continental
Thu, 5 December, 7:30 p.m.	Speech Communication	Regent
Thu, 5 December, 7:30 p.m.	Underwater Acoustics	Viceroy

STANDARDS COMMITTEES AND WORKING GROUPS

Mon, 2 December, 1:00 p.m.	S1.1 Standards Review (Acoustical Nomenclature)	Board
Mon, 2 December, 7:00 p.m.	ASACOS Steering	Windsor
Tue, 3 December, 7:30 a.m.	ASACOS	Kent/York
Tue, 3 December, 5:00 p.m.	WG44	Kent/York
Wed, 4 December, 10:00 a.m.	S1/WG9-Characterization of Windscreen Acoustical Performance	Board
Thu, 5 December, 10:00 a.m.	S1.1/WG9 ANSI S1.20 Calibration of Underwater Electroacoustic Transducers	Board

MEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Sun, 1 December 1:00 p.m. - 5:00 p.m. Mon, 2 December 8:00 a.m. - 12:30 p.m.	Short Course	Carousel
Mon-Thu, 2-5 December 7:30 a.m. - 5:00 p.m. Fri, 6 December 7:30 a.m. - 12:00 noon	Registration	Ballroom
Mon-Thu, 2-5 December, 7:00 a.m. - 5:00 p.m. Fri, 6 December, 7:00 a.m. - 12:00 noon	Internet Zone/E-mail	Grande Hall Foyer
Mon-Thu, 2-5 December, 7:00 a.m - 5:00 p.m. Fri, 6 December 7:00 a.m. - 12:00 noon	A/V Preview	Hospitality Suite 3101
Mon-Thu, 2-6 December 8:00 a.m. - 10:00 a.m. Mon, 2 December 9:45 a.m. - 11:00 a.m.	Accompanying Persons	Serea
Mon-Fri, 2-6 December, 10:00 a.m. - 11:00 a.m. Tue, 3 December 2:00 p.m. - 3:00 p.m.	Break	Ballroom/Grande Hall Foyer
Mon, 2 December 5:00 p.m. - 5:30 p.m.	New Student Orientation	Wilder
Mon, 2 December 5:30 p.m. - 7:00 p.m.	Student Meet and Greet	Vista Walk (O)
Mon, 2 December 5:30 p.m. - 7:00 p.m.	Exhibit Opening Reception	Ballroom
Tue, 3 December 9:00 a.m. - 5:00 p.m.	Exhibit	Ballroom
Tue, 3 December 7:00 a.m. - 8:00 a.m.	Yoga on the Beach	Main Beach (O)
Tue, 3 December 6:00 p.m. - 7:30 p.m.	Social Hour	North Beach (O)
Wed, 4 December 9:00 a.m. - 12:00 noon	Exhibit	Ballroom
Wed, 4 December, 11:45 a.m. - 1:45 p.m.	Women in Acoustics Luncheon	Vista Walk
Wed, 4 December 2:00 p.m. - 3:15 p.m.	Early Career Panel	Garden
Wed, 4 December, 4:15 p.m. - 5:30 p.m.	Plenary Session/Awards Ceremony	Upper Grande Hall
Wed, 4 December 5:30 p.m. - 7:00 p.m.	Hutchins Violin Quartet	Coronet
Wed, 4 December, 6:00 p.m. - 8:00 p.m.	Student Reception	Vista Walk (O)
Wed, 4 December, 8:00 p.m. - 12:00 midnight	ASA Jam	Regent
Thu, 5 December, 12:00 noon - 2:00 p.m.	Society Luncheon and Lecture	Ballroom
Thu, 5 December, 6:00 p.m. - 7:30 p.m.	Social Hour	Vista Walk (O)/Ballroom
Fri, 6 December 2:30 p.m. - 8:00 p.m.	Early-career Acousticians Retreat	Carousel
Sat, 7 December 8:00 a.m. - 1:00 p.m.	Early-career Acousticians Retreat	Carousel

178th Meeting of the Acoustical Society of America

The 178th meeting of the Acoustical Society of America will be held Monday through Friday, 2–6 December 2019 at The Hotel del Coronado, San Diego, California, USA.

SECTION HEADINGS

1. HOTEL INFORMATION

The Hotel del Coronado is the headquarters hotel where all meeting events will be held.

The cut-off date for reserving rooms at special rates has passed. Please contact The Hotel del Coronado, 1500 Orange Avenue, Coronado, CA 92118 (800-468-3533/619-435-6611) for information about room availability.

2. TRANSPORTATION AND TRAVEL

San Diego is served by the San Diego International Airport. Taxi fare to the Hotel del Coronado is approximately $30.

3. MESSAGES FOR ATTENDEES

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

4. REGISTRATION

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

Registration will open on Monday, 2 December, at 7:30 a.m. in the Ballroom (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 $660 for members of the Acoustical Society of America; $810 for non-members, $200 for Emeritus members (Emeritus status pre-approved by ASA), $380 for ASA Early Career members (for ASA members within three years of their most recent degrees – proof of date of degree required), $150 for ASA Student members, $250 for students who are not members of ASA, $25 for Undergraduate Students, and $200 for accompanying persons.

One-day registration is available at $380 for members and $455 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 $810 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 2020 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 $355 and includes a one-year Associate Membership in the ASA upon completion of an application form.

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

5. ACCESSIBLITY

If you have special accessibility requirements, please indicate this 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.

6. TECHNICAL SESSIONS

The technical program includes 129 sessions with 1251 abstracts scheduled for presentation during the meeting.

A floor plan of the Hotel del Coronado appears on page A11. Session Chairs have been instructed to adhere strictly to the printed time schedule, both to be fair to all speakers and to permit attendees to schedule moving from one session to another to hear specific papers. If an author is not present to deliver a lecture-style paper, the Session Chairs have been instructed either to call for additional discussion of papers already given or to declare a short recess so that subsequent papers are not given ahead of the designated times.

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

7. TECHNICAL SESSION DESIGNATIONS

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

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

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

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

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

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

8. HOT TOPICS SESSION

The Hot Topics session (3pID) will be held on Wednesday, 4 December, at 1:00 p.m. in the Hanover Room. Papers will be presented on current topics in the fields of Physical Acoustics, Biomedical Acoustics, and Computational Acoustics.

9. OTHER SPECIAL TECHNICAL EVENTS

A 3-hour hands-on workshop using an HITU SIMULATOR will be offered on Monday morning, 2 December, at 8:15 a.m. in the Coronet Room. This workshop is sponsored by the Biomedical Acoustics Technical Committee and will be available to all who are interested. There is no fee to participate, however, at-meeting registration is subject to availability of space in the workshop.

The Acoustical Society of America (ASA) seeks to engage and foster members by hosting the Early-Career Acousticians Retreat (EAR) 2019! EAR is a two-day workshop for early career professionals in the field of acoustics focused on developing leadership and networking skills for early career professionals in the field of acoustics. The workshop also will allow you to connect and socialize with your fellow early career acousticians as well as more senior members of the Society, learn about mentoring relationships and about the Society, and contribute to the future of ASA. Preregistration was required for this event.

The ASA is hosting a grant panel discussion designed to allow early career scholars to learn more about funding opportunities, which will be held on Wednesday, 4 December, from 2:00 p.m. to 3:15 p.m. in the Garden Room. Representatives from various granting agencies (e.g., NIH, NSF, ONR) as well as scholars experienced with navigating the funding landscape will provide information about grants geared towards early career scholars. Following the presentations, attendees will have opportunities to ask questions of the panelists. This event will be most relevant for those in the early career stage (e.g., post-doctoral fellows or assistant professors) or senior graduate students. If you have any questions about this event, contact Tessa Bent (tbent@indiana.edu) or Tyrone Porter (tmp@bu.edu).

ASA is partnering with the United States Jet Noise Reduction Science and Technology Panel to hold a workshop targeting emerging trends and technologies in jet noise solutions. The workshop will occur on Friday, 6 December, from 8:45 a.m. to 4:15 p.m. in the Continental/Crystal Room and is and open to all meeting registrants.

10. MEDWIN PRIZE IN ACOUSTICAL OCEANOGRAPHY AND ACOUSTICAL OCEANOGRAPHY PRIZE LECTURE

The 2019 Medwin Prize in Acoustical Oceanography will be presented to Dr. Chen-Fen Huang, Institute of Oceanography, National Taiwan University, at the Plenary Session on Wednesday, 4 December. Dr. Huang will present the Acoustical Oceanography Prize Lecture titled “The Development of Acoustic Mapping of Ocean Currents in Coastal Seas” on Tuesday, 3 December, at 11:00 a.m. in Session 2aAOb in the Empress Room.

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

The 2019 Rossing Prize in Acoustics Education will be presented to Preston S. Wilson, University of Texas at Austin, at the Plenary Session on Wednesday, 4 December. Dr. Wilson will present the Acoustics Education Prize Lecture titled “Learning and teaching acoustics through bubbles” on Tuesday, 3 December, at 11:00 a.m. in Session 2aEDb in the Carousel Room.

12. TUTORIAL ON EFFECTIVE MEDIA INTERACTIONS TRAINING WORKSHOP

The Public Relations Committee and the AIP Media Services team present this hands-on workshop for meeting attendees who are interested in effectively communicating scientific work to the public on Monday, 2 December, at 7:00 a.m. in the Regent Room. The workshop will consist of short presentations by media professionals to provide a toolkit of specific ideas and techniques for speaking to the media as well as structured small group activities that will give participants an opportunity to discuss and apply those techniques. Participants should come prepared to give a one-minute “elevator talk” about their own research.

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

13. SHORT COURSE ON ACOUSTOFLUIDICS

A short course on Acoustofluidics will be given in two parts: Sunday, 1 December, from 1:00 p.m. to 5:00 p.m. and Monday, 2 December, from 8:00 a.m. to 12:30 p.m. in the Carouse Room.

The instructor is James Friend, Professor at the Center for Medical Devices and Instrumentation in the Department of Mechanical and Aerospace Engineering, Jacobs School of Engineering, and the Department of Surgery, School of Medicine at the University of California, San Diego. Acoustofluidics involves acoustic wave propagation in solids and fluids and through interfaces; nonlinear effects including acoustic streaming and turbulence; bulk fluid, free surface, and suspended particle physical effects.

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

14. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday, Wednesday, and Thursday at The Hotel del Coronado. The schedule and rooms for each Committee meeting are given on page A16.

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.

15. EXHIBIT

An instrument and equipment exhibition will be located in the Ballroom near the registration area and meeting rooms and will open on Monday, 2 December, with an evening reception serving lite snacks and a complimentary drink. Exhibit hours are Monday, 2 December, 5:30 p.m. to 7:00 p.m., Tuesday, 3 December, 9:00 a.m. to 5:00 p.m., and Wednesday, 4 December, 9:00 a.m. to 12:00 noon.

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

Contact the Exhibit Manager for information about participating in ASA Exhibits: Dan Cooke, Director of Advertising and Exhibit Sales, AIP Publishing, LLC, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, Tel: 516-576-2629; E-mail: dcooke@aip.org.

16. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 4 December, at 4:15 p.m. in the Grande Hall.

ASA scholarship recipients will be introduced. The Medwin Prize in Acoustical Oceanography, the Rossing Prize in Acoustics Education, the A. B. Wood Medal and Prize, the Silver Medal in Musical Acoustics, and the Silver Medal in Physical Acoustics will be presented, Certificates will be presented to Fellows elected at the Louisville meeting. See page 2966 for a list of fellows and award recipients.

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

17. ANSI STANDARDS COMMITTEES

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

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

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

18. COFFEE BREAKS

Morning coffee breaks will be held each day from 9:45 a.m. to 11:00 a.m. in the Ballroom near the Exhibit and in the Grande Hall Foyer. On Tuesday there will be a coffee break in the Ballroom near the Exhibit from 2:30 p.m. to 3:30 p.m.

19. A/V PREVIEW ROOM

Hospitality Suite 3101 off the Garden Patio will be set up as an A/V preview room for authors’ convenience and will be available on Monday through Thursday from 7:00 a.m. to 5:00 p.m. and Friday from 7:00 a.m. to 12:00 noon.

20. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

The San Diego 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.

21. E-MAIL AND INTERNET ZONE

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

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

Tables with power cords will be set up in the Grande Hall Foyer for attendees to gather and to power-up their electronic devices.

22. SOCIALS

Socials will be held on Tuesday and Thursday evenings, 6:00 p.m. to 7:30 p.m. Tuesday’s social will be held on the Beach and Thursday’s social will be held at the Vista Walk (outdoor venue) and in the Ballroom.

The ASA hosts these social hours to provide a relaxing setting for meeting attendees to meet and mingle with their friends and colleagues as well as an opportunity for new members and first-time attendees to meet and introduce themselves to others in the field. A second goal of the socials is to provide a sufficient meal so that meeting attendees can attend the open meetings of Technical Committees that begin immediately after the socials

23. SOCIETY LUNCHEON AND LECTURE

The Society Luncheon and Lecture, sponsored by the College of Fellows, will be held Thursday, 5 December, at 12:00 noon in the Ballroom.

This luncheon is open to all attendees and their guests Purchase your tickets at the Registration Desk before 10:00 a.m. on Wednesday, 4 December. The cost is USD $30.00 per ticket.

24. STUDENT EVENTS: NEW STUDENTS ORIENTATION, MEET AND GREET, STUDENT RECEPTION

Follow the student twitter throughout the meeting @ ASAStudents,

A New Students Orientation will be held on Monday, 2 December, from 5:00 p.m. to 5:30 p.m. in Wilder. This will be followed by the Student Meet and Greet from 5:30 p.m. to 6:45 p.m. at the Vista Walk (outdoor venue) where refreshments and a cash bar will be available.

The Students’ Reception will be held on Wednesday, 4 December, from 6:00 p.m. to 8:00 p.m. at the Vista Walk (outdoor venue). 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.

25. WOMEN IN ACOUSTICS LUNCHEON

The Women in Acoustics luncheon will be held at 11:30 a.m. on Wednesday, 4 December, at Vista Walk (outside venue). Those who wish to attend must purchase their tickets in advance by 10:00 a.m. on Tuesday, 3 December. The fee is USD $30 for non-students and USD $15 for students.

26. JAM SESSION

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

27. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the San Diego meeting. The on-site registration fee for accompanying persons is USD $200. A hospitality room for accompanying persons will be open in the Serena Restaurant, 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.

28. WEATHER

San Diego experiences near perfect weather year-round. Average temperatures in December are highs of 66° F (19° C) and lows of 49° F (9° C). Chance of precipitation is about 2%.

29. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Aaron M. Thode, Technical Program Chair; Megan Ballard, Acoustical Oceanography; Jason Mulsow, Animal Bioacoustics; Shane Kanter, Benjamin Bridgewater, Architectural Acoustics; Kang Kim, Demi Libertario, Biomedical Acoustics; Jennifer Cooper, Computational Acoustics; Michael Haberman, Caleb Sieck, Engineering Acoustics; Daniel Russell, Education in Acoustics; Whitney Coyle, Peter Rucz, Musical Acoustics; William Murphy, James Phillips, Hales Swift; Noise; Kevin Lee, Physical Acoustics; Ellen Peng, Psychological and Physiological Acoustics; Ryan Harne, Kai Gemba, Signal Processing in Acoustics; Yoonjeong Lee, Susannah Lee, Rajka Smiljanic, Speech Communication; Benjamin Shafer, Anthony Bonomo, Robert M. Koch, Structural Acoustics and Vibration; Timothy Duda, Underwater Acoustics; Kieren Smith, Student Council.

30. MEETING ORGANIZING COMMITTEE

Peter Gerstoft, Chair; Aaron M. Thode, Technical Program Chair; Ludovic Tenorio-Hallé, Eric Snyder, Student Coordination; Gihoon Bryun, Signs; Michael Bianco, JAM; Sienna Thomas, Accompanying Persons; Camille Pagniello, Luncheons, Emma Ozanich, Fun Run

31. PHOTOGRAPHING AND RECORDING

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

32. ABSTRACT ERRATA

This meeting program is Part 2 of the October 2019 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*.

33. GUIDELINES FOR ORAL PRESENTATIONS

Preparation of Visual Aids

- See the guidelines for computer projection in section 41 below.
- Allow at least one minute of your talk for each slide (e.g., PowerPoint). No more than 12 slides for a 15-minute talk (with 3 minutes for questions and answers).
- Minimize the number of lines of text on one visual aid. 12 lines of text should be a maximum. Include no more than 2 graphs/plots/figures on a single slide. Generally, too little information is better than too much.
- Presentations should contain simple, legible text that is readable from the back of the room.
- Characters should be at least 0.25 inches (6.5 mm) in height to be legible when projected. A good rule of thumb is that text should be 20 point or larger (including labels in inserted graphics). Anything smaller is difficult to read.
- Make symbols at least 1/3 the height of a capital letter.
- For computer presentations, use all of the available screen area using landscape orientation with very thin margins. If your institutions logo must be included, place it at the bottom of the slide.
- Sans serif fonts (e.g., Arial, Calibri, and Helvetica) are much easier to read than serif fonts (e.g., Times New Roman) especially from afar. Avoid thin fonts (e.g., the horizontal bar of an e may be lost at low resolution thereby registering as a c.)
- Do not use underlining to emphasize text. It makes the text harder to read.
- All axes on figures should be labeled.
- No more than 3–5 major points per slide.
- Consistency across slides is desirable. Use the same background, font, font size, etc. across all slides.
- Use appropriate colors. Avoid complicated backgrounds and do not exceed four colors per slide. Backgrounds that change from dark to light and back again are difficult to read. Keep it simple.
- If using a dark background (dark blue works best), use white or yellow lettering. If you are preparing slides that may be printed to paper, a dark background is not appropriate.
- If using light backgrounds (white, off-white), use dark blue, dark brown or black lettering.
- DVDs should be in standard format.

Presentation

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

34. 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.

35. GUIDELINES FOR USE OF COMPUTER PROJECTION

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

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

Introduction

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

Guidelines

- Set your computer's screen resolution to 1024×768 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 1024×768 for an XGA projector or at least 640×480 for an older VGA projector. (1024×768 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.

36. 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

179th Meeting, Chicago, Illinois, 11–15 May 2020
180th Meeting, Cancun, Mexico, 9–13 November 2020
181st Meeting, TBD, spring 2021
182nd Meeting, Sydney, Australia, 6–10 December 2021

ASA School 2020

Living in the Acoustic Environment

9-10 May 2020
Itasca, IL

ASA School 2020 is an Acoustical Society of America event for graduate students and early career acousticians in all areas of acoustics to learn about and discuss a wide variety of topics related to the interdisciplinary theme *Living in the Acoustic Environment*. ASA School 2020 follows on the success of four previous ASA Schools starting in 2012, and will provide opportunities for meeting faculty and fellow students, mentoring, discussing research topics, and developing collaborations and professional relationships within acoustics.

Program and Costs

ASA School 2020 will take place at Eaglewood Resort and Spa in Itasca, IL, a scenic resort 30 minutes from Chicago, IL. Lectures and demonstrations followed by discussions will be given by distinguished acousticians in a two-day program covering topics in *architectural acoustics, animal bioacoustics, biomedical acoustics, engineering acoustics, musical acoustics, noise, psychological and physiological acoustics, and speech communication*. Although ASA School 2020 will focus primarily on these 8 technical areas, graduate students and early career professionals in all areas of acoustics are encouraged to attend to achieve a broader understanding of the diverse fields of acoustics.

The registration fee is $50. Hotel rooms at Eaglewood for two nights (double occupancy) and meals will be provided by ASA. Participants are responsible for their own travel costs and arrangements including transportation to Eaglewood. Transportation from Eaglewood to the ASA meeting location in Chicago at the close of ASA School 2020 will be provided and paid by ASA.

Participants and Requirements

ASA School 2020 is targeted to graduate students and early career acousticians (within 3 years of terminal degree) in all areas of acoustics. Attendance is limited to 60 participants who are expected to attend all School events and the ASA meeting immediately following on 11-15 May 2020. ASA School attendees are required to be an author or co-author on an abstract for presentation at the ASA Chicago meeting.

Application and Deadlines

The application form and preliminary program will be available online in November, 2019, at www.AcousticalSociety.org.

ASA
ACOUSTICAL SOCIETY OF AMERICA

Session 1aAA

Architectural Acoustics and Noise: Sustainable Acoustics for Smart Cities

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

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

Chair's Introduction—9:45

Invited Papers

9:50

1aAA1. Urban acoustic futures: Smart sources, paths and receivers. Glenn E. Sweitzer (none, 4504 N Hereford Dr., Muncie, IN 47304, glenn.sweitzer@gmail.com)

Climate changes will help inform, reimagine, reform, and even relocate urban areas. Higher ground, inland, will be more prized for habitation. Borders will be challenged—or erased. Building, transportation, communication, energy, governance, and even religious sectors will be challenged to provide services satisfactory to most. The staggering costs of new infrastructures will further exaggerate inequalities among still growing human populations. Specification of smart acoustic sources, transmission paths, and receiver conditions will nonetheless aid communication while mitigating annoyance. Lifelong learning will prove vital toward realizing and expressing individual versus collective acoustic needs. To explore potentials, alternative acoustic scenarios are developed and compared for carbon neutral urban areas for four extreme climate zones.

10:10

1aAA2. Innovative noise mitigation measures for traffic noise impact in Hong Kong. David B. Yeung (Ramboll Hong Kong Ltd., 21st BEA Harbour View Ctr., 56 Gloucester Rd., Wan Chai 00000, Hong Kong, dyeung@ramboll.com)

Hong Kong is a densely populated location with residential areas facing excessive traffic noise. Innovative measures in terms of Baffle Type and Plenum Acoustic Window have been invented, which design to improve the well-being from noise and building ventilation's aspect. To enhance the noise reduction performance of the acoustic windows, different absorption products like micro-perforated absorbers and perforated absorption panels had been tested. With diversified building layouts, designers are urged to optimize the window acoustic performance due to the forever changing flat dimensions due to market needs. These changes include increase in the overlapping length of the plenum of the window or by adding features to the façade of the plenum balcony door. Numerous window designs of different configurations have been evaluated with overlapping length, different acoustic treatment, and even outer window or balcony door design. To investigate the performance of the acoustic windows with different design parameters, a series of acoustic laboratory tests have been carried out in accordance with the ISO Standard to discover the relationship between the key design parameters of the window and its corresponding acoustic performance. The field tests were done in full scaled flat mock up using linear array of loud speakers mincing traffic noise source.

10:30

1aAA3. Reducing noise going into residential flats without closing windows—A new way of sustainable living format. K. K. Iu (East and South East Asia Regional Chapter, ASA, Hong Kong, Hong Kong), Maurice Yeung (East and South East Asia Regional Chapter, ASA, Macau, China), Andy Chung (East and South East Asia Regional Chapter, ASA, Hong Kong, Hong Kong), and Siu Kit Lau (East and South East Asia Regional Chapter, ASA, Block SDE3, #01-06, 4 Architecture Dr., Singapore 117566, Singapore, slau@acousticsresearch.com)

Accommodating over 7 million population in about 1100 km^2 area in which some 80% are hills, Hong Kong is truly a small and densely populated city, and housing demand is always a problem. Due to shortage of available sites, new residential buildings are inevitably constructed next to busy roads. It is very common that noise levels at flats are 10–12 dB(A) above the acceptable level adopted. "Innovative Acoustic windows" which consists of two layers of windows offer a good solution. The outer window is a push-pull type while inner window is a sliding window. Residents can position windows to reduce noise while maintaining ventilation. If the inner sliding window is aligned with the outer open window, more air and noise going indoor. If the sliding window is set offside of the outer window, it will reduce noise while still allowing air to circulate into the flat. Noise can be further reduced by using a noise absorption material to fit the window. In essence, "Innovative Acoustic windows" are capable to reduce 6 to 8 dB(A). This kind of "open window environment" is a preferred form of sustainable living format. This paper discusses how to evaluate noise reduction ability through laboratory test as well as the noise impact assessment process.

10:50

1aAA4. *Hibla*: An alternative sound absorption material. Shaira C. Gozun (Dept. of Education, Angeles City Sci. High School, Doña Aurora St., Lourdes Sur East, Angeles City 2009, Pampanga, Philippines, shaira.gozun@gmail.com), Neil David C. Cayanan, E'van Relle M. Tongol, and Lolita G. Bautista (Dept. of Education, Angeles City Sci. High School, Angeles City, Pampanga, Philippines)

Noise is a major concern in mechanical systems and poor acoustic facilities. Environmental and health issues of the commercially available acoustic materials led the researchers to develop comparable products from biomasses. This study utilized Abacá (*Musa textilis*), Bamboo (*Bambusa merrilliana*), and Water hyacinth (*Eichhornia crassipes*) that possess properties for sound absorption. These were extracted, blended with polyester (carrier fiber) in proportions of 50:50 and 25:75 (biomass-polyester), carded, and needle-punched. Abacá-Polyester 50:50 (0.106 SAA) performed best in Non-Standardized ASTM C423-17, surpassing the commercial Rockwool (0.058 SAA). Soundproof Test with Testo816-1 Decibel Meter resulted to 53.94 dB of Water hyacinth-Polyester 50:50 comparable to 53.93 dB of ambient noise (negative control) and 53.95 dB of Rockwool. ASTM E1050-12 Standardized Test Method for Impedance and Absorption of Acoustical Material validated the sound absorption property of all *Hibla* panels (50:50) with test results of 0.82 for Water hyacinth-polyester, 0.59 for Bamboo-polyester, and 0.58 for Abacá-polyester. As a non-woven fabric, *Hibla* passed the Standardized Tests of Thermogravimetric Analysis, Differential Thermal Analysis, Flammability Test, and Breaking Load in Tension Test. Thus, *Hibla* proved to have high sound absorption and thermal insulation, fire resistant, durability, and cost-efficient. Keywords: noise, acoustic, biomass, sound absorption, *Hibla* (fiber)

11:10

1aAA5. Landscape effects on soundscape experience in residential public spaces. Siu Kit Lau (Dept. of Architecture, National Univ. of SIngapore, Block SDE3, #01-06, 4 Architecture Dr., Singapore 117566, Singapore, slau@acousticsresearch.com), Saju Sajin, Wenluo Yu (Dept. of Architecture, National Univ. of SIngapore, Singapore, Singapore), Jian Kang (Univ. College London, London, United Kingdom), Shiu-Keung Tang, Chi Kwan Chau (The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), Stephen Siu Yu Lau (Dept. of Architecture, National Univ. of SIngapore, Singapore, Singapore), and Siew Eang Lee (Dept. of Bldg., National Univ. of Singapore, Singapore, Singapore)

There are limited studies on the audio-visual interactions in the soundscape environment for public spaces in residential premises. To fill the gap, an analysis of the effects of landscape elements on the soundscape perception of public spaces in Singapore's residential premises is performed. Two spaces are selected for the field questionnaire survey in accordance with the demographics of Singapore island. The residents of each locality are selected via random sampling for the survey. The questionnaire is developed and modified based on the recommendations of ISO 12913. The survey is formulated carefully to find the relationship between landscape characteristics and overall soundscape preference. Effects of individual landscape elements—visual landscape (landscape features that are constructed for visual appearance) and functional landscape(landscape elements that are constructed for a function or purpose)—are also analysed. The results show that there are differences between visual landscape and functional landscape effects on soundscape preferences, and more landscape elements should be considered as a better approach to create positive soundscape during public spaces design processes.

11:30

1aAA6. A study on impact of noise annoyance from highway traffic using noise map and structural equation model in Singapore City. Cheng S. Chin (Newcastle Univ., 537, #06-01, Clementi Rd., Singapore 599493, Singapore, cheng.chin@ncl.ac.uk) and Zi Yu Thang (Newcastle Univ., Singapore, Singapore)

This study uses traffic noise mapping and structural equation model to evaluate the impact of noise annoyance from highway traffic noise. The indices generated in noise maps by CityMap have verified the actual field measurements obtained from the site. The structural equation model addresses the relationship between noise annoyance and psychological factors. The results show that the noise annoyance is mainly shaped by the subject's concern and noise disturbance including the negativity on perceived disturbance and noise sensitivity. The noise levels are affected by the type of vehicles in the road, mainly on the heavy truck with more axles instead of the count of vehicles only. The traffic noise level can exceed the permissible limit established by a local government agency at a particular time of the day and on an overhead bridge situated near the vicinity of the residential area.

Session 1aAB

Animal Bioacoustics: General Topics in Animal Bioacoustics I

Alyssa W. Accomando, Chair
Biologic and Bioacoustic Research, National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste 200, San Diego, California 92106

Contributed Papers

9:00

1aAB1. A paradigm to explore bistatic matching in the bottlenose dolphin (*Tursiops truncatus*). Dorian S. Houser (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org), Jason Mulsow, Megan Tormey, Teri Wu, Leah Crafton, Rachel Simmons (National Marine Mammal Foundation, San Diego, CA), Mark J. Xitco, and James J. Finneran (US Navy Marine Mammal Program, San Diego, CA)

Match-to-sample (MTS) has been demonstrated in dolphins using echolocation and across sensory modalities (e.g., visual to echolocation and vice-versa). To assess potential for bistatic MTS, a pilot study was conducted in which a dolphin listened to one of three recorded playbacks of echoes obtained from targets of different shape and material composition and was then asked to echo locate on a set of physical alternatives and choose the corresponding target. Training began with two alternatives and 50% errorless trials, where only the matching physical target was presented. The proportion of errorless trials was progressively reduced to 25%, 12.5%, and then 0% as a criterion of three sessions with >80% correct matching was met. Thereafter, all three alternatives were presented with 0% errorless trials. The dolphin consistently performed above chance over 34 sessions—average correct matching rate was 76% (range 67%–96%)—even though the projected sample echoes did not faithfully reproduce echoes obtained when the dolphin insonified the targets with echolocation clicks. Future testing with novel stimulus sets would be required to determine if and how a dolphin might spontaneously generalize between projected echo facsimiles and real world targets.

9:15

1aAB2. Bistatic echo discrimination in the bottlenose dolphin (*Tursiops truncatus*). Katie Christman (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, katie.christman@nmmf.org), Danielle Ram (National Marine Mammal Foundation, San Diego, CA), Sean P. Coffinger (Univ. of California, San Diego, San Diego, CA), Jason Mulsow (National Marine Mammal Foundation, San Diego, CA), James J. Finneran (US Navy Marine Mammal Program, San Diego, CA), and Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA)

Prior research suggests that dolphins can echo-discriminate targets by eavesdropping on conspecific echolocation or by listening to the pulse-echo sequence of an electronic sonar system. To evaluate dolphins as bistatic receivers/classifiers, a three-year old, male bottlenose dolphin was trained to produce a conditioned phonic response upon hearing a "target" echo ($n = 1$) in a stream of "distractor" echoes ($n = 7$). Echoes, which were generated by insonification of variable shapes constructed of 1/8-inch thick stainless steel, were highly similar. Echoes were presented sequentially and grouped into "packets" of the same echo; three to nine packets of distractor echoes were randomly presented prior to presentation of the target echo packet. A 30-ms inter-echo interval was implemented and packet durations were randomized between 1.8 and 2.7 s. Echo amplitudes were first equalized with regard to their peak-peak pressures, then roved (± 3 dB) to eliminate amplitude cues. The dolphin's discrimination threshold was determined through an adaptive staircase procedure and defined as the echo level (calculated as the ratio of echo energy to the nominal within-band noise spectral density) corresponding to 50% correct discrimination. The dolphin's threshold of discrimination was determined to be $\sim$1 dB, which was similar to a prior report on dolphin passive echo discrimination.

9:30

1aAB3. Cognitive hierarchy of acoustic power spectrum features for simulated biosonar target echoes in the bottlenose dolphin (*Tursiops truncatus*). Alyssa W. Accomando (Biologic and BioAcoust. Res., National Marine Mammal Foundation, 2240 Shelter Island Dr., #200, San Diego, CA 92106, alyssa.accomando@nmmpfoundation.org), Jason Mulsow (Biologic and BioAcoust. Res., National Marine Mammal Foundation, San Diego, CA), Dorian S. Houser, and James J. Finneran (US Navy Marine Mammal Program, San Diego, CA)

Previous work with bottlenose dolphins [Dubrovsky *et al*., "Mechanisms of signal discrimination and identification in the auditory system of Tursiops truncatus," in *Marine Mammal Sensory Systems*, edited by Thomas *et al.* (Plenum, New York, 1992), pp. 235–240] suggested that the perception of coarse envelopes of echo power spectra ("macrostructure") is hierarchically dominant to finer-scale spectral features ("microstructure"). In the present study, two dolphins passively listened to and discriminated between two standard click doublets having different micro- and macrostructure. The dolphins were provided food reinforcement for remaining on an underwater station after the "negative" stimulus (two clicks with 100-μs separation) and for touching a paddle after the "positive" stimulus (two clicks with 150-μs separation). The dolphins were then presented with probe stimuli that were hybrids of the two standard stimuli. Each probe had a macrostructure identical to the positive or negative stimulus but the microstructure of the alternate standard. Preliminary results show the dolphins responding to probes in a manner consistent with macrostructure supremacy. This is in line with previous work suggesting that the macrostructure is hierarchically dominant over microstructure. [Work supported by the Office of Naval Research.]

9:45

1aAB4. Relationship between biosonar click emissions, age, and hearing bandwidth in bottlenose dolphins, *Tursiops truncatus*. Madelyn Strahan (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, madelyn.strahan@nmmf.org), James J. Finneran (NIWC Pacific Code 56710, US Navy Marine Mammal Program, San Diego, CA), Jason Mulsow, and Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA)

Previous studies have reported changes in biosonar emissions in a few odontocete subjects as audible frequency range is reduced with increasing age (i.e., presbycusis). For example, some animals have been observed to lower the dominant frequencies of their biosonar clicks to better match their reduced audible frequency range. In the present study, relationships between age, biosonar click emissions, auditory evoked potentials, and hearing

bandwidth were studied in 10 bottlenose dolphins (*Tursiops truncatus*) of various ages and hearing capabilities. Underwater hearing thresholds were estimated by measuring steady-state auditory evoked potentials to sinusoidal amplitude modulated tones at frequencies from 20 to 160 kHz. Input-output functions were generated at each tested frequency and used to calculate frequency-specific thresholds and the upper cut-off frequency of hearing for each subject. Click emissions were measured during a physical target aspect change detection task at a fixed range of 10 m. Relationships between hearing thresholds and the acoustic parameters of biosonar signals will be presented and compared to previous experiments with fewer subjects. [Work supported by the Office of Naval Research.]

10:00

1aAB5. Active infotaxis as a model for echolocation. Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu), John R. Buck (Elec. & Comput. Eng., Univ. of Massachusetts Dartmouth, North Dartmouth, MA), Peter L. Tyack (School of Biology, Univ. of St. Andrews, St. Andrews, Fife, United Kingdom), and Barbara Shinn-Cunningham (College of Eng., Carnegie Mellon Univ., Pittsburgh, PA)

Recent literature established the highly adaptive nature of echolocation: both toothed whales and bats modify their signals and sonar beam movements dynamically based on echoes from previous emissions in a task-depending manner. However, no theoretical framework systematically interprets these search processes across varying taxa and experimental conditions. To address this need, we cast the "infotaxis" model originally proposed for simulating moth odor tracking into the context of echolocation-based target search. Infotaxis posits that the animal chooses their next action to maximize the expected information gain. By doing so, the searching animal balances the exploration of new information about its environment with exploitation of existing information. The echolocator chooses the next sonar beam location and signal content based on its internal representations of the search space and sensory uncertainty. We model the echolocator's internal representation of the search space, the echo signatures of the target and clutter objects, and information acquired from echoes as probabilistic distributions and employ a Bayesian update of the target distribution after each echolocation reception. Simulations show the repeated inspection and switching among objects commonly observed in behavioral experiments of echolocation search. [Work supported by ONR MURI program.]

10:15–10:30 Break

10:30

1aAB6. Auditory brainstem responses to stimulus offset in the bottlenose dolphin (*Tursiops truncatus*). James J. Finneran (NIWC Pacific Code 56710, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Ryan A. Jones, Jason Mulsow (National Marine Mammal Foundation, San Diego, CA), and Robert F. Burkard (Univ. at Buffalo, Buffalo, NY)

The auditory brainstem response (ABR) is typically associated with the onset of a sound stimulus; however, ABRs can also be produced at sound offset. Here, ABRs in response to stimulus offset were examined in normal hearing (NH) and hearing impaired (HI) bottlenose dolphins. Tests were conducted in San Diego Bay, with the dolphin positioned underwater in front of a piezoelectric sound projector. Scalp activity was amplified (94 dB) and filtered (300–3000 Hz) before synchronous, weighted averaging and digitization. Each ABR was the averaged response to 512 stimuli; two ABRs were obtained for each stimulus condition. Stimuli included spectrally pink noisebursts (bandwidth: 20–160 kHz) and tonebursts (40-kHz; 113-kHz for the NH dolphins only). Stimulus level, risetime, and duration were systematically manipulated across experimental trials. Regardless of onset/offset envelope (cosine or linear), large amplitude offset responses were observed to 40-kHz tonebusts presented at higher levels with faster rise times for the NH but not the HI animals. One interpretation of these findings is that the offset response may be arising from more basal regions of the cochlea than the onset response. [Work supported by US Navy Living Marine Resources Program.]

10:45

1aAB7. Effects of jawphone position and stimulus frequency on the auditory brainstem response in bottlenose dolphins (*Tursiops truncatus*). Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, jason.mulsow@nmmf.org), Robert F. Burkard (Univ. at Buffalo, Buffalo, NY), Alyssa W. Accomando (National Marine Mammal Foundation, San Diego, CA), and James J. Finneran (US Navy Marine Mammal Program, San Diego, CA)

Suction-cups with embedded acoustic transmitters ("jawphones") can be used to deliver sound to odontocetes while out of water. When used for electrophysiological hearing tests, both jawphone and recording electrode location are important considerations. This study examined bottlenose dolphin auditory brainstem response (ABR) patterns across jawphone and electrode location. Stimuli were tonebursts with center frequencies of 28–160 kHz. When recorded using an electrode immediately behind the blowhole, ABR peak amplitudes were generally highest when the jawphone was placed on the middle of the mandible and lowest when placed near the tip of the rostrum. However, responses were comparable between the middle and posterior mandible when recorded with an electrode near the auditory meatus. These patterns were consistent across frequency, suggesting that ABRs recorded at the blowhole likely contain more binaural contributions than those recorded at the meatus. While blowhole electrode placements with a jawphone on the middle of the mandible provide the highest signal-to-noise ratios, they may not provide ear-specific information at suprathreshold stimulus levels, which could lead to confusion regarding the optimal location for transmitting sound to the ear. [Funded by US Fleet Forces Command.]

11:00

1aAB8. In-air and underwater auditory evoked cortical responses in the dolphin. Matt Schalles (Dept. of Cognit. & Neural Systems, Biomedical Eng., Boston Univ., 677 Beacon St., Boston, MA 02215, matt.schalles@gmail.com), Jason Mulsow, Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA), James J. Finneran (US Navy Marine Mammal Program, San Diego, CA), and Barbara Shinn-Cunningham (Carnegie Mellon Univ., Pittsburgh, PA)

Auditory signal processing in dolphins is of special interest because these animals echolocate and possess high upper-cutoffs of hearing (~140 kHz). The few previous measurements of auditory cortical responses (ACRs) in dolphins have been performed in air or with a restrained dolphin half submerged in water. Our goal was to record ACRs from an unrestrained dolphin fully submerged in water utilizing far-field sound stimulation. We performed a series of experiments in air and under water manipulating stimulus presentation rate and level. An increasing stimulus rate from 2 to 4 Hz showed only a nominal decrease in ACR potential; a greater decrease in ACR amplitude occurred with an 8-Hz rate. Decreasing stimulus sound pressure level (SPL) decreased the ACR in a manner consistent with auditory brainstem responses to similar SPLs. Taken together, these findings support the idea that valid cortical potentials can be obtained in air and underwater, though differences in ACR amplitude and polarity require further investigation. Brainstem responses are consistent between both environments; however, the cortical response is attenuated underwater, and a polarity reversal along the anterior-posterior axis observed in air is not present underwater.

11:15

1aAB9. Designing a waveguide to transmit sound to a dolphin in a functional magnetic response imaging machine. Jawanza K. Foster (Elec. Eng., UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, jwzfoster@gmail.com) and John R. Buck (Elec. Eng., UMass Dartmouth, North Dartmouth, MA)

Functional Magnetic Resonance Imaging (fMRI) offers great opportunities to learn how a dolphin's brain processes acoustic echolocation and communication signals. Delivering sound to an animal in a fMRI requires a sound delivery system with no ferrous metal. This talk proposes an approach that generates sound far from the fMRI magnet and then conducts the sound through a waveguide to the dolphin roughly 10 m away. The waveguide functions at two frequency ranges are 80 kHz for echolocation signals and 10 kHz to 20 kHz for communication signals. This approach has two challenges: will there be enough energy for the dolphin to hear the acoustic signals and will the received signal be distorted? The waveguide loses energy due to the radiation impedance around the hydrophone, absorption and friction, and the impedance mismatch between the waveguide and the dolphin. Filling the waveguide with water limits the transmission loss through the waveguide. The waveguide radius will be chosen to limit the number of trapped modes, minimizing distortion due to dispersion. [Funded by the ONR MURI program.]

11:30

1aAB10. Acoustic characteristics near the living pool of captive belugas whale. Zhengliang Cao (Shanghai Ocean Univ., 999 Hucheng Huan Rd., Shanghai 201306, China, caozhengliang@yahoo.com), Liming Song, and Zhong Chen (Shanghai Ocean Univ., Shanghai, China)

Belugas are widely known for their diverse and frequent calls and have been given the nickname, “sea canaries.” As the number of belugas has been living in aquariums and ocean parks, it is very important for us to provide a comfortable environment for these creatures. That means we should understand how sounds made by belugas vocalization and environmental noise related to belugas habitats, therefore designing the most suitable artificial pool for these creatures. This study is on sound acquisition and acoustic estimation near the living pool of captive belugas whales in a new Ocean Park. Besides different hydrophones using for passive acoustic recording, accelerometer sensors and sound meters are deployed to obtain structural vibrations and airborne noises, respectively. Acoustic signals in different sites and pools were recorded, and their characteristics are analyzed and compared to mainly understand the relationship between underwater acoustics and structural vibrations. In the future, more detail tests will design to find out what is the main noise source which may be influencing belugas’ living. This study is also helpful to apply appropriate and reasonable methods for noise reduction. [Work supported by National Natural Science Foundation of China (Grant No. 41374147).]

MONDAY MORNING, 2 DECEMBER 2019 EMPRESS, 8:15 A.M. TO 11:20 A.M.

Session 1aAO

Acoustical Oceanography and ASA Committee on Standards: Observational Acoustical Oceanography: A Look at Enabling Technology from Academia and Industry I

Andrey K. Morozov, Cochair
Marine, Teledyne, 49 Edgerton Drive, North Falmouth, Massachusetts 02556

Orest Diachok, Cochair
Johns Hopkins University APL, 11100 Johns Hopkins Rd., Laurel, Maryland 20723

Chair’s Introduction—8:15

Invited Papers

8:20

1aAO1. Advances in technology and signal processing for ocean acoustic tomography. Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225, pworcester@ucsd.edu) and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

The acoustic source and receiver technologies used to make large-scale tomographic measurements have advanced significantly in recent years. High-efficiency, swept-frequency sources in the 200–300 band have been developed by Teledyne Webb Research. Ultra-low frequency sources in the 30–40 Hz band required for measurements in ice-covered regions have been developed by GeoSpectrum Technologies, Inc. Acoustic receiver technology has also improved. Early instruments were complicated devices because of the scientific requirements for precision timekeeping, measurement of the motion of the moored instruments, and the storage of large amounts of acoustic data. Modern developments in data acquisition systems and data storage have now made the required instrumentation more user friendly. Distributed Vertical Line Array (DVLA) receivers made up of distributed, self-recording Hydrophone Modules and a small number of central controllers now allow large vertical receiving arrays to be deployed. Of equal importance to the hardware developments, the processing of tomographic data has become much more routine. Estimator-correlator processing explicitly accounts for scattering in the receptions. The Viterbi algorithm is used for automated peak tracking to obtain time series of travel times. Finally, travel times (and other data) are now being used to directly constrain ocean circulation models to estimate the state of the ocean.

8:45

1aAO2. *In situ* measurements of compressional and shear wave properties during gravity coring operations. Megan S. Ballard (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), Kevin M. Lee, Andrew R. McNeese, and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Sediment cores provide valuable insight about the physical properties of the seabed, and laboratory measurements of sediment wave speed from cores are often considered "ground truth." However, sound-speed estimates obtained from cores can be inaccurate due to changes in pressure, temperature, and mechanical properties of the sediment caused by the removal of the core from the seabed and its subsequent transport to the laboratory. To address these issues, the Acoustic Coring System (ACS) was developed. The system uses sets of transducers mounted below the penetrating tip of a sediment corer to make *in situ* measurements of geoacoustic properties as the corer penetrates the seabed. Compressional wave measurements are obtained with rod-mounted piezoelectric cylinders, and shear wave measurements are obtained with bender elements mounted in flat blades. The ACS was deployed during the environmental survey for ONR Seabed Characterization Experiment (SBCEX) in 2016 and in support of the ONR Canada Basin Acoustic Propagation Experiment (CANAPE) in 2017. In both cases, the system provided measurements of the sediment sound-speed profile as a function of depth within the seabed. On-going work focused on the measurement of shear wave speed profiles and compressional and shear wave attenuation will also be presented. [Work supported by ONR.]

9:10

1aAO3. Acoustical ocean ecology in the era of the robot revolution. Kelly J. Benoit-Bird (Monterey Bay Aquarium Res. Inst., 7700 Sandholdt Rd., Moss Landing, CA 95039, kbb@mbari.org)

Over the past decade, the field of oceanography has experienced a robotic revolution. A proliferation of autonomous surface and underwater platforms is providing new ways to obtain environmental, chemical, and biological data. Bioacoustical oceanographic sensors are increasingly being added to the payload of these platforms to provide increased spatial and temporal coverage as well as access to unexplored areas. Integrating relatively large, high power draw sensors like scientific echosounders is a challenge, however, and each platform type and size have specific tradeoffs with respect to these payloads. I will present case studies employing echsounders from autonomous surface vessels to study the deep scattering layer, a deep-diving, short endurance autonomous underwater vehicle for examining bathypelagic squid, and a moderate endurance underwater glider for examining the relationship between forage species and their environment. These studies illustrate the challenges of integration of echsosounder payloads into platforms as well as the opportunities of robotic approaches for bioacoustical oceanography. Finally, a recent experiment integrating nine robotic platforms and a cabled observatory will highlight how the integration of various sensor technologies and platforms can be used to study dynamic biological processes like diel vertical migration in the ocean.

Contributed Papers

9:35

1aAO4. Operational ocean acoustic tomography. Alison B. Laferriere (Raytheon BBN Technologies, 10 Moulton St., Cambridge, MA 02138, alison.laferriere@raytheon.com), Pierre F. Lermusiaux (Massachusetts Inst. of Technol., Cambridge, MA), Yevgeniy Dorfman, Michael Goldsmith (Raytheon BBN Technologies, Cambridge, MA), Kevin D. Heaney (Appl. Ocean Sci., Fairfax Station, VA), and Aaron Kofford (Raytheon BBN Technologies, Arlington, VA)

The estimation of sound speed fields using ocean acoustic tomography (OAT) in operational systems is limited because user intensive post processing and manual association of arrivals with ray paths are required. Furthermore, inversion techniques typically rely on CTD casts or historical data to create reference sound speed profiles and Gaussian statistical characterizations for the inversion, which may be either unavailable, out of date, or too approximate due to the intermittent ocean dynamics. A method is presented for real-time implementation of OAT, including the tracking of acoustic arrivals, model-data association of ray paths to arrivals (ray identification), and tomographic inversions. One- and two-dimensional tomographic inversions for sound speed fields are performed using stochastic ensemble forecasts of the sound speed field generated via the Multidisciplinary Simulation, Estimation, and Assimilation Systems (MSEAS) Primitive-Equation probabilistic ocean modeling system. The method is demonstrated on simulated and real data obtained from moored tomographic sources and receivers, deployed in deep water 100 km South of Nantucket, MA. [This research was developed with funding from the Defense Advanced Research Projects Agency (DARPA).]

9:50–10:05 Break

10:05

1aAO5. Underwater glider localization using broadband source transmissions 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), Matthew Dzieciuch, and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Two Seaglider autonomous underwater vehicles were deployed as mobile receiving platforms in a 2016–2017 tomography experiment in the Canada Basin. The introduction of gliders into a tomography experiment has the advantage of adding acoustic data receptions at many depths and ranges relative to moored acoustic sources. The challenge in interpreting these data is the lack of statistics at any given location and the uncertainty in the glider position during a dive, the latter resulting in a fundamental ambiguity between the position and sound speed. The acoustic arrival matching localization technique, performed in post-processing, has been employed to position the instruments while underwater using transmissions from moored acoustic sources. This technique capitalizes on the broadband nature of the tomography sources and estimates latitude and longitude while relying upon the vehicle's pressure sensor for depth. Acoustic ranging was also performed onboard the gliders during the deployment, and estimated ranges from the moored sources were transmitted via Iridium satellite link every few hours when the glider surfaced. Positions resulting from acoustic arrival matching localization will be compared with results based on semi-real time acoustic ranging. Results will be discussed in the context of reviewing the state of the art in underwater glider localization.

10:20

1aAO6. A real-time acoustic drifter for active and passive sonar applications. Altan Turgut (Naval Res. Lab, Acoust. Div., Code 7160, Washington, DC 20375, altan.turgut@nrl.navy.mil) and Jeffrey Schindall (Naval Res. Lab, Washington, DC)

A low-cost, long-duration acoustic drifter has been developed to measure directional ambient noise and man-made acoustic signals in both shallow and deep oceans. Traditional acoustic and oceanographic measurements are typically performed by deploying fixed moorings that would limit retrieving the data only after the recovery. With the advancements in low-power CPUs and satellite telemetry, real-time oceanographic and acoustic measurements are possible with in-buoy data reduction and satellite transmission of processed data. A Raspberry Pi computer with a 16-channel data acquisition board is used for both active and passive sonar applications, including directional ambient measurements, seabed characterization, and ocean acoustic tomography. A specific version of the acoustic drifter was built for on-ice deployments in the Arctic as an ice-tethered acoustic buoy. Several examples of its applications from recent sea-going experiments are given. Successful deployments and instant data analysis would indicate a practical alternative to more traditional acoustic and oceanographic measurement techniques by providing a real-time monitoring capability. [Work supported by the ONR.]

10:35

1aAO7. Advances in marine vibrators for seismic survey. Andrey K. Morozov (Marine, Teledyne, 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com)

The reducing impact of noise from traditional air-guns on marine mammals is an actual problem for oil and gas producers. The application of coherent, broadband signals instead of explosive, high intensity, and air-gun pulses is the potential way of noise mitigation. The three major oil companies, Shell, Exxon, and Total, are sponsoring the Marine Vibrator Joint Industry Project MVJIP. Marine Vibrators are a coherent type of seismic source, which are less harmful for marine inhabitants and give a clearer, higher resolution imaging of the sub-bottom formations. Teledyne as one of the participants in the MVJIP using a large underwater bubble resonator as a new type of seismic source. The cylindrical bubble keeps its shape in the water and can be towed with the speed 8 knots. The polyurethane membrane covering the gas-filled bubble supports a high volume displacement. The research shows that the best driver for ultra-low frequency 1–10 Hz source is the blower with the airflow controlled by a proportion valves. The experimental prototype has good performance with a maximum more than SPL 200 dB at the frequency 5–12 Hz. However, for a limited sound pressure level of 185 dB in a frequency band of 10–100 Hz, the subwoofer driver shows very good performance.

10:50

1aAO8. Azimuthal, spatial, and temporal variability of acoustic intensity in cross-shelf direction during the yearlong shallow water Canada Basin Acoustic Experiment 2016–2017. Mohsen Badiey (Univ. of Delaware, Newark, DE 19716, badiey@udel.edu), Lin Wan (Univ. of Delaware, Newark, DE), Sean Pecknold (DRDC - Atlantic, Dartmouth, NS, Canada), and Altan Turgut (Acoust., Naval Res. Lab., Washington, DC)

A yearlong study of spatial, temporal, and azimuthal variability of sound propagation with simultaneously measured oceanography on Chukchi shelf is reported. In a shallow water region, two acoustic sources were deployed for studying the "along" and "cross-shelf" propagation. The "along-shelf" study is presented separately [*J. Ascout. Sci. Am.* **145** (2019)]. Here, we focus on the "cross-shelf" signal propagation in two frequencies (0.7–1.1 and 1.5–4 kHz) transmitted from a single sound source placed near the sound channel axis in 320 m water depth. Three "cross-shelf" acoustic tracks connected the source and three receiver arrays placed along 50 m isobath. The angle between east most and west most tracks was around 106 deg. Another array at 250 m isobath was deployed along the middle track. Sound emitted from the common source shows different behavior along each track. Concurrently, detailed water column salinity and temperature were measured by environmental arrays in both "along" and "cross-shelf" directions. Sea surface ice was measured by an upward looking sonar for several months. Seasonal injection of different water masses in this region and variations of sea surface conditions (full-ice, transition, and free-surface) are examined using the acoustical oceanographic data. This paper quantifies analysis using correlation between acoustics and oceanographic signals. [Work supported by ONR 322OA.]

11:05

1aAO9. Long range intensity fluctuations on the Chukchi continental shelf measured during the year-long Canada Basin Acoustic Propagation Experiment 2017. Mohsen Badiey (Univ. of Delaware, Newark, DE 19716, badiey@udel.edu), Lin Wan, Christian D. Escobar (Univ. of Delaware, Newark, DE), Sean Pecknold (DRDC - Atlantic, Dartmouth, NS, Canada), Richard A. Krishfield, Andrey Y. Proshutinsky (Woods Hole Oceanographic Inst., Woods Hole, MA), John A. Colosi (Naval Post Graduate School, Monterey, CA), Peter F. Worcester, and Mathew Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA)

Analysis of one-year-long broadband transmissions from five deep water sound sources recorded on shelf during the Canada Basin Acoustic Propagation shows strong azimuthal variability. Broadband chirp signals (140–325 Hz) transmitted every 4 h were received on spatially distributed receiver arrays on the Chukchi shelf from October 2016 through 2017. Here, we present spatial, temporal, and azimuthal correlations by four receiver arrays with source-receiver distances ranging from 228.6 to 524.4 km in a cross-the-shelf orientation. Temporal fluctuations in received field are related to water column variability for each source-receiver pair. Received signals from the same source to each receiver on the shelf, separated from 15.8 to 49.63 km in "along-the-shelf" direction show large intensity fluctuation. This is related to the turbulence that signals have experienced in their path. Temporal and spatial variations of temperature profiles were measured on the shelf, however, from deep water sources to the shelf break region, there is little information. Analysis are conducted for three time zones, from fully frozen seas, transition from full ice, and ice-free conditions. The parabolic equation model is used to report the propagation of field for the ice-free condition. Limitations of the acoustic modeling due to spatially under-sampled input environmental parameters are discussed. [Work supported by ONR-322 OA.]

MONDAY MORNING, 2 DECEMBER 2019 CORONET, 8:15 A.M. TO 11:15 A.M.

Session 1aBAa

Biomedical Acoustics: Ultrasound Modeling Workshop—HITU Simulator

Joshua Soneson, Chair
Applied Mechanics, FDA, 10903 New Hampshire Ave., Silver Spring, Maryland 20993

This two-hour hands-on workshop uses the recently updated HITU_SimulatorV2.0. HITU_Simulator is a free, user-friendly, Matlab-based package for finite-amplitude axisymmetric fields which uses a wide-angle KZK-type propagation model and includes thermal modeling and dose prediction.

MONDAY MORNING, 2 DECEMBER 2019 HANOVER, 10:00 A.M. TO 11:45 A.M.

Session 1aBAb

Biomedical Acoustics and Physical Acoustics: Cavitation Nuclei—Bubbles, Droplets, and More I

James J. Kwan, Cochair
Institute of Biomedical Engineering, University of Oxford, Old Road Campus Research Building, Oxford, Oxfordshire, United Kingdom

Shashank Sirsi, Cochair
Bioengineering, UT Dallas, 800 West Cambell Rd., Dallas, Texas 75080

Invited Papers

10:00

1aBAb1. Photoacoustic imaging with nanoparticle contrast media. Jesse V. Jokerst (NanoEng., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0448, jjokerst@ucsd.edu)

Photoacoustic imaging combines the high temporal and spatial resolution of ultrasound with the good contrast and spectral nature of optics. This techniques is "light in/sound out" as opposed to traditional "sound in/sound out" ultrasound. In this lecture, I will present three case studies that highlight the power of photoacoustic imaging to address the needs of the medical community. First, I will describe our work using photoacoustics to guide therapy in treating multidrug-resistant bacteria. Second, I will discuss our efforts in oral health including improved transducers for minimally invasive exams. Finally, I will detail photoacoustic imaging with a wearable transducer for therapeutic drug monitoring of heparin.

10:20

1aBAb2. Horses for courses (or something much smaller!): The design, development and acoustic testing of sub-micron cavitation nucleation agents for biomedical applications. Christophoros Mannaris, Luca Bau (Dept. of Eng. Sci., Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), James J. Kwan (School of Chemical and Biomedical Eng., Nanyang Technol. Univ., Singapore, Singapore), Boon Teo (School of Chemistry, Monash Univ., Melbourne, Victoria, Australia), Michael Gray, Delphine Elbes, Cameron Smith, Catherine Paverd, Robert Carlisle, Eleanor P. Stride (Dept. of Eng. Sci., Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), and Constantin Coussios (Dept. of Eng. Sci., Inst. of Biomedical Eng., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, constantin.coussios@eng.ox.ac.uk)

Lipid and protein-shelled microbubbles have been the mainstay of cavitation nucleation strategies for therapeutic applications for several decades. However, a growing number of therapeutic ultrasound applications require nuclei that (i) are significantly smaller in size, in order to overcome a particular biological barrier such as the leaky vasculature of tumours or the stratum corneum; (ii) offer greatly increased cavitation persistence, both during a single extended ultrasound pulse and in terms of extended circulation following intravenous administration; and (iii) have better resilience to sudden ambient pressure changes in order to enable direct injection without nuclei destruction into tissue targets via a needle and syringe. We will review a range of novel cavitation agents currently under development, including sub-micron gas-stabilizing solid particles and nanodroplets, and provide an overview of their known characteristics in terms of acoustic emissions, activation and cavitation thresholds, cavitation persistence, and circulation. Where possible, the relevant performance of these cavitation nucleation agents will be compared to microbubbles for applications ranging from drug delivery to nucleated tissue fractionation.

10:40

1aBAb3. Lipid-functionalized porous oxide nanoparticles for acoustic imaging and site-directed therapy. Andrew P. Goodwin (Chemical and Biological Eng., Univ. of Colorado Boulder, 9500 Gilman Dr., M.S. 0448, La Jolla, CA 92093, agoodwin@ucsd.edu)

While most US contrast agents are fluid-filled micro- or nanoparticles, their stability of fluid-filled particles is limited by inherent Laplace pressure and equilibrium with the surroundings. Here, I will present our work designing silica nanoparticles of $\sim$100 nm with specifically tailored surfaces that can nucleate the formation of ultrasound-responsive microbubbles under reduced acoustic pressures. I will discuss the underlying materials science behind their activity, the effects of chemical functionalization on acoustic contrast (with the potential for stimulus-responsiveness), and applications in imaging and directed mechanical cell killing.

Contributed Papers

11:00

1aBAb4. Targeted ultrasound enhanced drug delivery to foam cell spheroids. Reju G. Thomas (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., 62 Nanyang Dr., Singapore 637459, Singapore, gtreju@ntu.edu.sg), Xiaoqian Su (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., Singapore, Singapore), Catarina Vizetto-Duarte, Aristo Muktabar, Kee W. Ng (School of Mater. Sci. and Eng., Nanyang Technolog. Univ., Singapore, Singapore), and James J. Kwan (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., Singapore, Singapore)

Atherosclerosis is a chronic vascular disorder marked by the accumulation of foam cells in the intimal space of the artery. Foam cells are macrophages that reside in the arterial wall after internalizing excessive cholesterol. Though many strategies to remove cholesterol from foam cells have been put forward, delivering these agents at the site of the lesion remains a challenge. Here, we investigated the ability of our previously reported multi-cavity poly-*co*-lactic-*co*-glycolic acid microparticles (mc-PLGA MPs) to penetrate into a model for early stage atherosclerosis, a 3D foam cell spheroid, and deliver hydroxyl beta cyclodextrin (HBCD) and sirolimus. HBCD is a polysaccharide that solubilizes cholesterol and promotes cholesterol efflux from foam cells. First, the drug release capability of mcPLGA MPs was tested using DAPI dye as a model drug. Sustained release of DAPI from mcPLGA MPs was observed in foam cell nuclei three days after implantation. We found that DAPI released from mcPLGA MPs distributed evenly throughout the spheroid even though mcPLGA MPs were implanted predominantly at the periphery. FITC conjugated HBCD was co-administered with sirolimus loaded mc-PLGA MPs and was also found to be implanted only under ultrasound exposure. Foam cell viability and the inflammatory response of the ultrasound treatment were also monitored across a week. In conclusion, this study shows the possibility of atherosclerosis treatment with targeted ultrasound enhanced drug delivery.

11:15

1aBAb5. Cavitation response from multi-cavity and porous PLGA microparticles. Xiaoqian Su (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., 62 Nanyang Dr., Block N1.2, B3-13, Singapore 637459, Singapore, S160010@e.ntu.edu.sg), Reju G. Thomas, and James J. Kwan (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., Singapore, Singapore)

Solid cavitation agents have become an emerging technology for both diagnostic and therapeutic ultrasound applications. The key trait amongst the many formulations of solid cavitation agents is the ability to trap gas either within pores or cavities throughout the particle. Recently, we have reported on the development of a degradable multi-cavity PLGA microparticle for ultrasound-enhanced drug delivery. The cavitation signal from these particles suggested the presence of both inertial and non-inertial cavitation. Here, we conducted a deeper analysis on the cavitation signal from both these multi-cavity PLGA microparticles as well as highly porous PLGA microparticles. We investigated their acoustic response to a spherically focused ultrasound transducer at various pressure amplitudes and frequencies. The power spectral density curves from the received acoustic signals, as measured by a passive cavitation detector, were obtained and analyzed for harmonic and broadband content. Ultrasound contrast enhancement using a conventional ultrasound imaging system of both porous and multi-cavity particles was also investigated. These results provide further insights into the design and acoustic response of solid polymeric cavitation agents for diagnostic and therapeutic applications.

11:30

1aBAb6. Controlled drug delivery and release in brain tumors with focused ultrasound. Costas Arvanitis (Mech. Eng. and Biomedical Eng., Georgia Inst. of Technol., 901 Atlantic Dr. NW, Rm. 4100Q, Atlanta, GA 30318, costas.arvanitis@gatech.edu), Yutong Guo, and Chulyong Kim (Mech. Eng. and Biomedical Eng., Georgia Inst. of Technol., Atlanta, GA)

Several passive and active strategies based on different nanoparticle (NP) formulations have been proposed for improving drug delivery in brain tumors. Despite improved NP designs, drug accumulation in brain tumors remains very low. In this talk, we will present two different strategies to improve the penetration of therapeutic agents in brain tumors. The first uses transcranial focused ultrasound (FUS) in combination with microbubbles to overcome the vascular barriers of brain tumors. Our findings indicate that ultrasonically actuated microbubbles can increase the interstitial convective transport ($Pe_{non\text{-}FUS} \approx 0.1$ to $Pe_{FUS} \approx 20$), in addition to alleviating the vascular barriers, and improve the extravasation of fluorescently labeled NPs ($\sim$40 nm in diameter) in healthy mice brain and mice brain tumor models. The second strategy employs transcranial MR guided focused ultrasound (MRgFUS) combined with low temperature sensitive liposomes loaded with doxorubicin (LTS-Dox $\sim$ 100 nm in diameter). Using an optimized MRgFUS system and closed-loop methods for controlled transcranial hyperthermia ($\sim$10 min), we were able to significantly increase doxorubicin uptake in LTS-Dox + FUS group as compared to LTS-Dox and free Dox groups ($<$0.01) in mice brain tumor model. Our results demonstrate that these therapeutic strategies provide unique opportunities to improve the delivery of NPs and their cargo in the brain and brain tumor microenvironment.

MONDAY MORNING, 2 DECEMBER 2019 CONTINENTAL/CRYSTAL, 9:15 A.M. TO 11:45 A.M.

Session 1aNS

Noise and Physical Acoustics: Quiet Supersonic Flights 2018 I

Jonathan Rathsam, Cochair
NASA Langley Research Center, MS 463, Hampton, Virginia 23681

Larry J. Cliatt, Cochair
NASA Dryden Flight Research Center, P.O. Box 273, Mail Stop 2228, Edwards, California 93523

Chair's Introduction—9:15

Invited Papers

9:20

1aNS1. Overview of Quiet Supersonic Flights 2018 (QSF18) in Galveston, Texas. Jonathan Rathsam (NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, jonathan.rathsam@nasa.gov) and Larry J. Cliatt (NASA Armstrong Flight Res. Ctr., Edwards, CA)

NASA and partners planned and executed the Quiet Supersonic Flights 2018 (QSF18) test program in Galveston, TX. The primary goal was to successfully engage a community, unaccustomed to hearing sonic booms, in a multiweek test of quiet supersonic flights. Secondary goals included assessing community survey methods and methods for sonic boom exposure determination. This presentation provides an overview of test planning, test execution, data analysis methods, dose-response results, and lessons learned. All of these elements inform future community response testing with NASA's X-59 Quiet Supersonic Technology (QueSST) aircraft.

9:40

1aNS2. Updates to PCBoom sonic boom propagation code: Modeling QSF18 sonic thumps. Joel B. Lonzaga (Structural Acoust. Branch, NASA Langley Res. Ctr., 2 N. Dryden St., B1208 MS 463, Hampton, VA 23681, joel.b.lonzaga@nasa.gov)

NASA's PCBoom sonic boom propagation code has recently been updated to improve the code's numerical accuracy and efficiency. This paper specifically discusses algorithms for calculating the ray tube area, accounting for turbulence effects, and determining the entire sonic boom carpet. Unlike most existing sonic boom propagation codes that require three or four neighboring acoustic rays for the ray tube area calculation, the updated PCBoom requires only one ray to perform the same calculation. In addition, the turbulence algorithm employed here is based on a multiple-scattering theory rather than based on a one-way parabolic equation used by existing algorithms. Numerical models obtained using the updated PCBoom are compared with a measured dataset from the Quiet Supersonic Flights 2018 research campaign (QSF18) that NASA conducted in Texas in 2018. The dataset was obtained using dive maneuvers performed by supersonic conventional aircraft that generated small-amplitude sonic booms, called sonic thumps, mostly with small elevation angles of arrival. In such sonic booms, the effects due to turbulence and spreading from refraction are large, making the dataset suitable for assessing the code's performance to accurately predict such effects.

10:00

1aNS3. Making sonic boom measurements during adverse weather: Summarizing BYU's participation in QSF18. Kent L. Gee (Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu), Daniel J. Novakovich, Reese D. Rasband, Kevin M. Leete (Brigham Young Univ., Provo, UT), and Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

One challenge with outdoor sonic boom measurements is making high-fidelity recordings under adverse weather conditions. This paper describes the deployment of a "weather-robust" measurement system as part of Brigham Young University's (BYU) participation in the NASA Quiet Supersonic Flights 2018 (QSF18) test in Galveston, TX. BYU made measurements at four different stations across the city representing potentially different ambient environments: Scholes airport, a cemetery, a city park near a busy street, and the U.S. post office. At each station, multiple measurements were made using different data acquisition hardware and microphone types and configurations. These recordings, made within a few meters of each other, allowed for the comparison of different measurement approaches in terms of practicality, instrumentation noise floor, measurement bandwidth, wind noise, etc. The test itself and data analysis demonstrated the viability of the weather-robust measurement system, under real-world adverse weather conditions. They also yielded some lessons learned and recommended paths forward for future sonic boom measurements. [Work supported by NASA Langley Research Center through the National Institute of Aerospace.]

10:20–10:35 Break

Invited Papers

10:35

1aNS4. Bayesian statistical dose-response models. Jasme Lee (National Inst. of Aerosp., NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, jasme.lee@nasa.gov), Jonathan Rathsam (NASA Langley Res. Ctr., Hampton, VA), and Alyson G. Wilson (North Carolina State Univ., Raleigh, NC)

In the coming years, NASA will collect nationally representative data to support efforts to develop international standards for permissible noise from commercial supersonic flight over land. The data will elucidate the relationship between sonic boom noise exposure and community response. This presentation introduces several Bayesian statistical models and a method to calculate a summary dose-response curve. Additionally, the statistical models illuminate the trade-off between sample size and precision of estimated quantities such as percent highly annoyed. This trade-off will inform sample size decisions for future community surveys. All models are fit to data from a recent NASA community response test, the Quiet Supersonic Flights 2018 (QSF18) test.

10:55

1aNS5. NASA supersonic testing: Streets, suburbs, and sonic booms. Matt Kamlet (Public Affairs, NASA, 25433 Via Nautica, Valencia, CA 91355, matthew.r.kamlet@nasa.gov)

NASA's Quiet Supersonic Flights 2018 campaign in Galveston, TX, or QSF18 for short, marked a major milestone in the agency's efforts to demonstrate quiet supersonicflight. The flights, which took place from 5 to 15 November 2018, marked NASA's first supersonic flight research series to feature active participation and feedback from a large community unaccustomed to quiet supersonic overflights. The data obtained fromQSF18 are directly preparing NASA for the Low-Boom Flight Demonstration mission, in which NASA's X-59 Quiet SuperSonic Technology X-plane, or QueSST, will fly overcommunities around the country to demonstrate the ability to reduce the loud, disruptive sonic boom typically associated with supersonic flight, to a quiet thump. NASA's efforts to communicate with the community played a critical role in the successor QSF18, just as it will play a critical role in X-59's Low-Boom Flight Demonstration mission. The process to successful communications, based on experience and lessons learned from previous supersonic flight research, is an effort that spans multiple fronts, including high public awareness, extensive communication based on each region's unique features, character, and resources, local government partnership, media coordination, educational resources, direct public communication and accommodation,and as much coordination as possible through local and regional emergency response organizations.

Contributed Papers

11:15

1aNS6. Impact of microphone configuration on sonic boom recordings. Daniel J. Novakovich (Dept. Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, danielnovakovich@gmail.com), Kent L. Gee, Reese D. Rasband (Brigham Young Univ., Provo, UT), and Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

Because different measurement setups can affect sonic boom recordings, this paper compares different microphone configurations employed during NASA's Quiet Supersonic Flights 2018(QSF18) research campaign in Galveston, TX. Hardware was deployed at two geographically unique locations that reasonably represented the three different measurement approaches across QSF18. Analysis of several booms points to acoustically relevant differences in the various configurations, most likely caused by microphone height and windscreen type. An elevated microphone has measurable multipath interference effects (also observed in complementary laboratory measurements) and is subjected to increased wind. Additionally, sensors with more compact wind screens have less wind-noise rejection. The multipath effects, in particular, can cause bias errors in relevant metric calculations. Thus, this comparison is useful in the design and planning of similar, future measurements. [Work supported by NASA Langley Research Center through the National Institute of Aerospace.]

11:30

1aNS7. Employing digital pole-shifting filters to improve low-frequency response of sonic boom measurements. Reese D. Rasband (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, r.rasband18@gmail.com), Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Thomas B. Gabrielson (Appl. Res. Lab., Penn State Univ., State College, PA), and Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

High-fidelity measurement of sonic boom waveforms requires microphones and data acquisition hardware with flat frequency responses that extend well below 1 Hz. However, digital pole-shifting filters that account for the microphone and data acquisition system roll-offs of conventional hardware can be used in post-processing to extend the effective bandwidth. This approach is demonstrated for sonic boom recordings from the NASA Quiet Supersonic Flights 2018 (QSF18) measurement campaign. Recordings of several booms at multiple measurement sites using different data acquisition hardware and microphone combinations were used to understand the robustness of the processing technique. In particular, manufacturers report nominal hardware roll-off frequencies with which filters can be designed. However, the QSF18 data can be used to derive filters computationally using a mean-square error method. Results, including residual errors and other limitations, are discussed. The transferability of the technique is further described through application to a launch vehicle reentry sonic boom, which had markedly different characteristics than the QSF18 recordings. [Work supported by NASA Langley Research Center through the National Institute of Aerospace.]

MONDAY MORNING, 2 DECEMBER 2019 WILDER, 9:00 A.M. TO 11:45 A.M.

Session 1aPA

Physical Acoustics: General Topics in Physical Acoustics I

Charles Thompson, Chair
ECE, UMASS, 1 University Ave., Lowell, Massachusetts 01854

Contributed Papers

9:00

1aPA1. Acoustic streaming near a sharp edge. Charles Thompson (ECE, UMASS, 1 University Ave., Lowell, MA 01854, charles_thompson@uml.edu), Kavitha Chandra, Arielle Joasil, and Sarah Kamal (ECE, UMASS, Lowell, MA)

This work examines the generation of acoustic streaming near a sharp edge. For an edge having an exterior angle larger than 180 deg, the local fluid velocity at its apex is singular. The implication of this state of affairs and the role it plays in vorticity generation and transport are examined. A theoretical model for the scattering and dissipation will be presented.

9:15

1aPA2. Acoustic radiation force and torque on inhomogeneous scatterers in the Born approximation. Thomas S. Jerome (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX 78713-8029, tsjerome@utexas.edu) and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

If a scatterer and the surrounding medium have similar material properties and the dominant contribution to the acoustic radiation force is due to energy density gradients in the sound field, the Born approximation may be used to reduce calculation of the radiation force and torque on a scatterer of arbitrary shape to an integral over the volume of the scatterer. Comparison with full solutions based on wave function expansions demonstrates that the approximation is accurate for homogeneous spheres and prolate spheroids in a standing plane wave when the size of the scatterer is less than about one wavelength [Jerome *et al.*, *J. Acoust. Soc. Am.* **145**, 36 (2019)]. Here, the Born approximation is extended to inhomogeneous scatterers by allowing the compressibility and density to vary within a scatterer in a standing plane wave. Analytical expressions for both the force and torque are obtained for a finite cylinder with inhomogeneity that varies linearly along its axis. Effects of inhomogeneity are examined for both cylinders and prolate spheroids. Physical insight is provided by simplified analytical expressions for narrow axisymmetric scatterers, such as thin rods, that are much smaller than a wavelength. [TSJ was supported by the ARL:UT McKinney Fellowship in Acoustics.]

9:30

1aPA3. Aligning high-aspect ratio particles in user-specified orientations with ultrasound directed self-assembly. Milo Prisbrey (Mech. Eng., Univ. of Utah, 201 Presidents Circle, Salt Lake City, UT 84119, mprisim@gmail.com) and Bart Raeymaekers (Mech. Eng., Univ. of Utah, Salt Lake City, UT)

We use ultrasound directed self-assembly (DSA) to create two-dimensional patterns of high aspect ratio particles with user-specified orientation. We theoretically derive a method to determine the operating parameters of any arrangement of ultrasound transducers, required to align high aspect ratio particles in any user-specified orientation. The method finds the ultrasound wave field that maximizes the curvature of the acoustic radiation potential orthogonal to the user-specified particle orientation and in user-specified locations. We simulate the theoretical solution resulting from this method and experimentally validate it using carbon microfibers in water, and we quantify the position and orientation error. The method enables controlling the location and orientation of high aspect ratio particles, including simultaneously orienting multiple high aspect ratio particles in different directions. This work finds application in the biomedical field and in using ultrasound DSA as a processing or manufacturing method for engineered materials.

9:45

1aPA4. Cell agglomeration using guided surface acoustic waves through a couplant layer. Jiyang Mei (Mech. and Aerosp. Eng., Univ. of California, San Diego, Matthews Lane, Rm 345C, Structural and Material Eng. Bldg., La Jolla, CA 92093, j1mei@eng.ucsd.edu), Kenjiro Takemura (Dept. of Mech. Eng., Keio Univ., Yokohama, Kanagawa, Japan), and James Friend (Mech. and Aerosp. Eng., Univ. of California, San Diego, La Jolla, CA)

Cell agglomeration has been useful and crucial for tissue engineering, cell culturing, and drug testing, such as for cancer drug development. However, most of the methods require specific and complex experimental setups and the resulting agglomerates are generally slow or of poor quality. We propose a microfluidic device using surface acoustic waves (SAWs), a contact-free means, in the application of cell engineering. It is composed of a piezoelectric substrate, 127.86 deg Y-rotated X-propagating lithium niobate, and a focused interdigital transducer to generate focused surface waves with a resonant frequency at 100 MHz. An aluminum guiding layer is deposited on top of the substrate to overcome the beam steering and lateral diffraction problems due to the anisotropy nature of the piezoelectric material, further trapping the wave to a small region of a few wavelengths. The SAWs generated travel into a thin layer of liquid and diffract at the Rayleigh angle, so that the acoustic radiation is coupled and propagates into the superstrate, which can be a bio-friendly container culturing cells. The localized acoustic streaming induced from the radiation carries and accumulates the cells to the center of the recirculation. Our data show that the agglomeration can form in less than 20 s, and the separation between the living yeast cell agglomeration islands can be as small as $\triangle x = 720\ \mu$m. This device is proven to be an effective, reliable, and easy-to-handle approach for cell manipulation, independent of the culturing container.

10:00

1aPA5. The effect of acoustically driven fluid motion on boundary-induced pattern formation. Charles Thompson (ECE, UMASS, 1 University Ave., Lowell, MA 01854, charles_thompson@uml.edu), Kavitha Chandra, and Flore Norude (ECE, UMASS, Lowell, MA)

This work examines the process of boundary-induced pattern formation in reaction-diffusion systems. The influence of acoustically induced time-averaged advection that results from the gradient of the Reynolds stress is of particular interest. The presentation will focus on the role that boundary vibration and unsteady fluid motion have on manipulating chemical concentration at the microscale.

10:15–10:30 Break

10:30

1aPA6. Acoustic localization using Green's function retrieval methods and range migration processing. Max Denis (U.S. Army Res. Lab., 1 University Ave., Lowell, MA 01854, max_f_denis@hotmail.com), Sandra L. Collier, John Noble, W. C. Kirkpatrick Alberts, David Ligon, Leng Sim, and Deryck D. James (U.S. Army Res. Lab., Adelphi, MD)

In this work, Green's function retrieval and frequency-wavenumber methods are employed to enhance array plane-wave beamforming maps for acoustic source localization and range estimation in an outdoor environment. The crosscorrelation and multidimensional deconvolution Green's function retrieval methods are used to improve the signal-to-noise ratio of the beamforming maps. The Stolt's frequency-wavenumber migration method is adapted to the plane-wave beamforming map to find the source positions, applying frequency-wavenumber migration image to the Stolt spatial transformation mapping. Open field microphone array measurements of active and passive sources are investigated. Of particular interests are the accuracy of the estimated source position, the effects of multiple sources, and the image contrast of the beamforming maps.

10:45

1aPA7. Towards 3-D thermoacoustic plasma confinement. Seth Pree (Univ. of California Los Angeles, 475 Portola Plaza, Los Angeles, CA 90095, sethpree@ucla.edu), John P. Koulakis, and Seth Putterman (Univ. of California Los Angeles, Los Angeles, CA)

Standing wave acoustic fields can segregate partially ionized gas by temperature via a generalized acoustic radiation pressure that we have called the pycnoclinic acoustic force. Thus far, these sound fields have been excited and sustained with a microwave source pulsed near the resonance frequency of a cavity. Consideration of the temperature and luminosity oscillations due to the adiabatic compression of a sound wave suggests an alternative method of driving the sound field necessary for confinement. Acoustic temperature oscillations in the presence of continuous (i.e., not pulsed) microwave fields may cause variable microwave absorption in phase with the acoustic oscillation so as to add energy to the sound field. If the energy added by microwave absorption exceeds that lost to acoustic damping, amplification, and possibly self-oscillation will occur. We give theoretical criteria for amplification and present the apparatus and measurements intended to find signatures of plasma thermoacoustics.

11:00

1aPA8. Imaging the shear and secondary compression wave: Ultrafast ultrasound in saturated foams reveals porous dispersion. Johannes Aichele (Labtau, INSERM Labtau - Univ. of Lyon, Cours Albert Thomas 152, Lyon 69003, France, johannes.aichele@inserm.fr), Bruno Giammarinaro (Labtau, INSERM Labtau - Univ. of Lyon, Lyon Cedex 03, France), Michael Reinwald (Bimedical Eng. - King's College London, Paris, France), Goulven LeMoign (Univ. of Lyon - Creatis, Lyon, France), and Stefan Catheline (Labtau, INSERM Labtau - Univ. of Lyon, Lyon, France)

Wave propagation in porous materials is of relevance in many fields of acoustics such as geophysics, noise cancellation, and biomedical imaging. In contrast to classical elastic materials, poroelastic materials support three types of elastic waves and exhibit a distinctive dispersion in the presence of viscous fluids. In addition to the compression and shear wave, a secondary compression wave, often named Biot slow wave, exists. Both, the slow compression wave and the shear wave are highly attenuated. This poses crucial difficulties for experimental detection. We overcome this challenge by using high frame rate ultrasound imaging for wave tracking inside saturated, highly porous melamine foams. To our knowledge, we show the first experimental speed and attenuation measurements inside a soft porous materials. In particular, experimental detection of the slow compression wave is scarce, and no direct imaging inside a porous material has been reported. Both wavespeeds are governed by the weak frame of the foam and exhibit a strong dispersion due to the fluid viscosity. Our experiments have direct implications for medical imaging: Melamine foams exhibit a similar microstructure as lung tissue. Furthermore, other organs such as the liver can be modeled as a soft porous material.

11:15

1aPA9. Acoustics of a piezo inkjet channel with an entrained air bubble. Tim Segers (Phys. of Fluids, Univ. of Twente, The Netherlands, Phys. of Fluids Group, TechMed Ctr., Postbus 217, Enschede 7500 AE, The Netherlands, t.j.segers@utwente.nl), Arjan Fraters (Phys. of Fluids, Univ. of Twente, The Netherlands, Enschede, The Netherlands), Hans Reinten, Youri de Loore (Océ Technologies B.V., Venlo, The Netherlands), Detlef Lohse, and Michel Versluis (Phys. of Fluids, Univ. of Twente, The Netherlands, Enschede, The Netherlands)

Piezo-acoustic drop-on-demand (DOD) inkjet printing is widely applied in high-end digital printing due to its unprecedented precision and reproducibility. However, the stability of piezo-DOD inkjet printing can sometimes be compromised through the stochastic entrainment of bubbles within the ink channel. The acoustically driven air bubble modifies the ink channel acoustics, and conversely, the modified ink channel acoustics influences the bubble dynamics. Here, we measure the acoustic eigenfrequency of a MEMS based silicon ink channel as a function of the bubble size. The eigenfrequency was measured using a pulse-echo system and the bubble

size using a short-wave infrared imaging setup. We show that the measured eigenfrequency increases when an air bubble is entrained. Surprisingly, the ink channel resonance frequency plateaus at total bubble volumes larger than 10 pl. Moreover, it was found that at a constant total bubble volume, the resonance frequency increases with the number of entrained bubbles. We show that both experimental observations can be quantitatively explained from a simple lumped element model (LEM) comprising the ink channel Helmholtz resonator coupled to the bubble mass-spring system. The results of the LEM model were validated using a full numerical model of the coupled ink channel—bubble system.

11:30

1aPA10. Evaluation of optoacoustic response with a combined Monte Carlo optical and acoustic simulation approach. Matthew W. Urban (Dept. of Radiology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu), James A. Rose, Benjamin Buhrow, and Clifton Haider (Dept. of Physiol. and Biomedical Eng., Mayo Clinic, Rochester, MN)

Optoacoustic, or photoacoustic, imaging combines the penetration capabilities of ultrasound imaging with the contrast mechanism of optical absorption related to the photoacoustic effect. To enable modeling of photoacoustic measurements and imaging applications, the problem can be divided into modeling of the optical photon propagation and the resulting acoustic wave propagation. In highly scattering media such as soft tissues, Monte Carlo (MC) methods are used to evaluate photon propagation, absorption and the subsequent generation of a heat source that produces the photoacoustic effect. Acoustic wave equations are then used to model the propagation of the sound in the medium. However, if the appropriate physics could be modeled with a single tool, there would be advantages of consistency in spatial and temporal domains as well as easier integration of the optical and acoustic simulations. We propose using MC methods for optical photon propagation and for the acoustic wave propagation. The simulations are split into two stages for the optical photon propagation and the acoustic phonon propagation. We describe how the same MC framework is used to model photon propagation and acoustic phonon propagation. We will then demonstrate this new combined MC framework in models of photoacoustic problems in homogeneous and heterogeneous cases. These results will be compared with results from using *k*-Wave models for the acoustic wave propagation. The correspondence between *k*-Wave the acoustic MC model are shown to be in good agreement.

MONDAY MORNING, 2 DECEMBER 2019 — SPRECKLES, 8:45 A.M. TO 11:45 A.M.

Session 1aSA

Structural Acoustics and Vibration: Acoustics of 3-D-Printed Materials and Structures

Alexey S. Titovich, Cochair
Naval Surface Warfare Center, Carderock Division, 9500 MacArthur Blvd, West Bethesda, Maryland 20817

Stephanie G. Konarski, Cochair
Naval Research Laboratory, 4555 Overlook Ave. SW, Washington, DC 20375

Chair's Introduction—8:45

Invited Papers

8:50

1aSA1. Additive manufacturing and architected materials: New process developments and materials. Christopher Spadaccini (Eng. Directorate, Lawrence Livermore National Lab., 7000 East Ave., Livermore, CA 94551, spadaccini2@llnl.gov)

Material properties are governed by the chemical composition and spatial arrangement of constituent elements at multiple length-scales. This fundamentally limits material properties with respect to each other creating trade-offs when selecting materials for specific applications. For example, strength and density are inherently linked so that, in general, the more dense the material, the stronger it is in bulk form. We are combining advanced design method such as topology optimization, with advanced additive micro- and nanomanufacturing techniques to create new material systems with previously unachievable property combinations – mechanical metamaterials. The performance of these materials is fundamentally controlled by geometry at multiple length-scales rather than chemical composition alone. We have demonstrated designer properties of these mechanical metamaterials in polymers, metals, ceramics and combinations thereof. Properties include ultra-stiff lightweight materials, negative stiffness, and negative thermal expansion to name a few, as well as functional properties such electrical, optical, acoustic, and chemical responses. We have primarily utilized our custom developed additive micro-manufacturing techniques to create these structures and materials. These include projection microstereolithography, direct ink writing, electrophoretic deposition, volumetric AM via tomographic reconstruction, parallel two-photon polymerization, and diode-based additive manufacturing of metals.

9:10

1aSA2. Design exploration of additively manufactured metamaterials. Carolyn C. Seepersad (Mech. Eng., The Univ. of Texas at Austin, 204 Dean Keeton, Austin, TX 78705, ccseepersad@mail.utexas.edu), Michael R. Haberman, and Clinton B. Morris (Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Additive manufacturing (AM) is making a profound impact on our ability to realize complex metamaterials and structures, but fully realizing the manufacturing freedom afforded by AM requires some significant advances in engineering design methods and tools. For some additive manufacturing applications, simulation-based design tools may be required to explore a hierarchy of features, ranging in size from microns to meters. At the same time, these tools need to provide real-time feedback on the constraints and process-structure-property relationships relevant to specific AM technologies, and this Design-for-AM feedback is needed *during* the design process, rather than at the end. To address these challenges, a design exploration approach has been established for creating inverse maps of promising regions of a hierarchical structural/material design space. The approach utilizes machine learning classifiers for identifying sets of promising solutions to a materials design problem, combined with statistical characterization of geometric features and material properties to ensure that the designs are robustly manufacturable. The capabilities of the approach are demonstrated by applying it to the hierarchical design of negative stiffness metamaterials for energy absorption applications

9:30

1aSA3. Effects of metal 3-D printing processes on acoustic metamaterial elements. Charles Rohde (Naval Res. Lab., Washington, DC, charles.rohde@nrl.navy.mil), Christina J. Naify, Andrew Birnbaum, and Matthew D. Guild (Naval Res. Lab., Washington, DC)

Acoustic metamaterial properties derive from inherently complex, sub-wavelength geometries which aim to control the propagation of sound and elastic waves. The material properties of the unit cell element, the *meta-atom*, are important when considering either resonant response, or when the transport medium is a high density fluid, such as water. For water applications, this typically means using metals in the design. Metal additive manufacturing (AM) is a logical approach to achieve the needed design complexity. The most common approach in metal AM is selective laser melting. This approach utilizes a high-power laser beam to locally melt powder into a solid. 3D geometries are built through layer-by-layer melting of material. This approach to fabrication incorporates thermodynamic complexities such as extreme temperature gradients, simultaneous multiple material phases, gradient driven convection, rapid solidification, and occasional void formation. We will outline the complications that arise when using this technology to build one of the simplest acoustic *meta-atoms*: a thin disk. We show that thermal stresses can thermally warp the geometry, and microstructural heterogeneities can locally change material properties. These effects alter both the AM membrane's macroscale (resonance) and microscale (crystal structure), significantly impacting the acoustic response. [Work Sponsored by the Office Of Naval Research.]

Contributed Papers

9:50

1aSA4. Three-dimensional printing of tunable stiffness acoustic materials. Christina J. Naify (Acoust. Div., Naval Res. Lab, 4800 Oak Grove Dr, Pasadena, CA 91109, christina.naify@gmail.com), Charles Rohde, Alec Ikei, and Caleb F. Sieck (Acoust. Div., Naval Res. Lab, Washington, DC)

Three-dimensional printing has opened up a wide range of new avenues in acoustics with the ability to rapid-prototype and to create structures with unusual acoustic properties such as space-coiling and exotic lattices. In most cases, however, the mechanical, and thus acoustic properties of the printed structures are fixed once printed. Recently, specifically within the area of acoustic metamaterials, the field of tunable acoustic structures have gained a large amount of interest for their ability to adapt to a wide range of frequencies. In this work, we explore the intersection of tunable acoustic structures and additive manufacturing by 3-D printing a structure whose acoustic response can be locally, actively tuned via resistive heating in order to decrease the elastic modulus of the material and achieve a more highly damped structure. This phenomena is demonstrated by tuning the acoustic response of membrane structures since many acoustic metamaterial phenomena such as sound insulation, negative density, negative index, supercoupling, acoustic steering, metasurfaces, and demonstration of Willis coupling, utilize membrane structures. Specifically, we present two demonstrations of tunable membranes to achieve non-symmetric acoustic modes in a normal incidence sound tube, and to tune Willis coupling in an extremely subwavelength unit cell. [Work Sponsored by the Office Of Naval Research.]

10:05

1aSA5. Utilizing a spark gap generator as an impulsive noise source for scale model experiments investigating novel metamaterial arrays. Gordon M. Ochi (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., 2902 Newmark Dr., Champaign, IL 61822, Gordon.M.Ochi@erdc.dren.mil), Michelle E. Swearingen, Megan Kreiger (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., Champaign, IL), Kyle G. Dunn, Michael Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Engineer and Res. Development Ctr., Hanover, NH), Emanuel J. Vargas (Construction Eng. Res. Lab., U.S. Army Engineer Res. and Development Ctr., Champaign, IL), and Douglas A. Punt (Cold Regions Res. and Eng. Lab., U.S. Army Engineer and Res. Development Ctr., Hanover, NH)

A spark gap generator is demonstrated as a surrogate blast source for scale model experiments. It provides a portable, repeatable, and safe option for investigating the propagation of impulsive signals through a 3-D printed metamaterial. Key insights into proper design and implementation of the spark gap circuit, relating to the characteristic frequency and peak pressure of the signal created by the spark gap, are given. Properties of the signal, including directivity, duration, and spectral characteristics, are discussed. The ability of the spark gap to replicate features of an explosives related shock wave, such as the rise time and N-wave behavior, is also discussed. Finally, the spark gap generator is used to conduct scale model experiments with novel 3-D printed metamaterial arrays, which are compared to simulated results.

10:20–10:35 Break

Invited Papers

10:35

1aSA6. Acoustics education through digital fabrication. John Granzow (Performing Arts Technol., Univ. of Michigan, 660 Lomita Dr., Stanford, CA 94305, jgranzow@umich.edu)

The integration of makerspaces in academic settings presents an opportunity to apply principles of musical acoustics to the rapid production of sounding objects. Digital fabrication accelerates the design cycle and alters what can be accomplished in the scope of such educational labs. Acoustic principles motivate parametric 3-D designs using open-source CAD software. These models can then be materialized through digital fabrication to empirically test numerical predictions; they also serve as prototypes for new instrument design and sound art. This paper reviews such a course offered at the University of Michigan in the Performing Arts Technology program. Machine and software requirements are discussed as well as methods to combine acoustic problem-sets with contemporary fabrication methods.

10:55

1aSA7. Integrated *in situ* and *ex situ* ultrasonic characterization of Ti6Al4V parts made with directed energy deposition additive manufacturing with hybrid capabilities. Joseph A. Turner (Dept. of Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, W342 Nebraska Hall, Lincoln, NE 68588-0526, jaturner@unl.edu), Luz D. Sotelo, Rakeshkumar Karunakaran, Cody J. Kanger, and Michael P. Sealy (Dept. of Mech. and Mater. Eng., Univ. of Nebraska-Lincoln, Lincoln, NE)

Metal additive manufacturing (AM) is employed to make highly complex low volume components, which often have demanding performance requirements. Current challenges in the metal AM field include control of porosity and microstructure, development of process-structure-property relationships, and simplification of the AM parameter space. These challenges are exacerbated in multiphase materials, such as Ti6Al4V, that are important for the biomedical, aerospace, and energy industries. Integrated *in situ* and *ex situ* non-destructive evaluation methods are proposed to address these challenges. In this work, *in situ* and *ex situ* ultrasound measurements are conducted on Ti6Al4V parts made using a hybrid Directed Energy Deposition (DED) system. The hybrid capabilities of the system are exploited to quantify part geometry, while ultrasound is used to measure phase velocity and attenuation *in situ* on a layer-by-layer basis. Furthermore, the phase velocity, attenuation, and backscatter responses of these AM and Hybrid AM parts are quantified *ex situ* and compared with those from conventionally manufactured Ti6Al4V parts to evaluate the role of microstructural complexity. The results highlight the improved geometric information offered by a hybrid AM system to quantify microstructure and elastic properties of parts *in situ*. The need for realistic complex microstructures to be integrated within model-based ultrasonic microstructural predictions is also discussed.

Contributed Papers

11:15

1aSA8. Acoustic signals associated with laser-substrate interaction in powder bed fusion additive manufacturing process. Robert W. Smith (Appl. Res. Lab., Penn State, P.O. Box 30, State College, PA 16804, rws100@psu.edu), David J. Corbin, Jan Petrich, and Edward W. Reutzel (Appl. Res. Lab., Penn State, State College, PA)

During the Powder Bed Fusion Additive Manufacturing Process, flaws are sometimes produced in the component. It is desirable to detect these flaws during the deposition process, so that they might be remediated before they become captured within the part. A pilot study was conducted to see if acoustic data might show some signature or variation when such flaws are produced. This paper will provide some observations on acoustic data recorded during such an additive manufacturing deposition process. Acoustic data were collected within the chamber during the Powder Bed Fusion Additive Manufacturing Process in a 3D Systems Launches Pro-X 320 Direct Metal Printer. A PCB 1/4 inch model 378A14 pre-polarized piezo-electric pressure microphone with an integrated preamplifier was the sensor used to detect acoustic signals. Tonal noise associated with pumps obscured the spectral region below 10 kHz, but significant acoustic output associated with the laser-substrate interaction was present from this lower limit. A small cylinder was produced for test, during which intentional process variations were introduced. A CT scan of the cylinder was used to locate flaws, and to register these locations, and thus to attempt to correlate these flaw locations with time windows in the acoustic data.

11:30

1aSA9. Ultrasonic non-destructive characterization of hybrid additively manufactured 420 stainless steel made with directed energy deposition. Luz D. Sotelo (Mech. and Mater. Eng., Univ. of Nebraska - Lincoln, 900 N 16th St., W 342 NH, Lincoln, NE 68588-0526, luz.sotelo@huskers.unl.edu), Cody S. Pratt, Haitham Hadidi, Michael P. Sealy, and Joseph A. Turner (Mech. and Mater. Eng., Univ. of Nebraska - Lincoln, Lincoln, NE)

Additive manufacturing is remarkably suitable for creating high complexity performance parts. An added benefit of the Directed Energy Deposition (DED) process is the ability to make functionally graded materials. This expansion of the design and manufacturing spaces presents a challenge for the non-destructive evaluation of AM parts. Hybrid AM parts are an example of functionally graded materials for which the variation in microstructure and material properties across the bulk of the parts is created through a combination of manufacturing processes. In this work, a hybrid 420 stainless steel coupon was created using the DED method, and a surface treatment was applied between added layers. Characterization using destructive and non-destructive methods was performed. The microstructure, hardness profile, phase velocity, attenuation, and backscatter results from the hybrid coupon are compared with those from an AM coupon and a wrought coupon. Agreement between destructive and non-destructive measurements is studied. Furthermore, ultrasonic non-destructive methods are shown to be effective for identifying the gradient in the material properties of the hybrid coupon. The work hereby presented can further inform non-destructive evaluation decisions for hybrid AM parts.

Session 1aSC

Speech Communication: Universal and Experiential Influences on Phonetic Perception

Linda Polka, Cochair
School of Communication Sciences & Disorders, McGill University, 2001 McGill College Avenue, 8th floor, SCSD, McGill University, Montreal, Quebec H3A IGI, Canada

Matthew Masapollo, Cochair
Cognitive, Linguistic & Psychological Science, Brown University, 677 Beacon St., Boston, Massachusetts 02215

Chair's Introduction—9:00

Invited Papers

9:05

1aSC1. Acoustic versus articulatory accounts of asymmetries in vowel perception. Matthew Masapollo (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., 677 Beacon St., Boston, MA 02215, mmasapol@bu.edu)

Most research on cross-language speech perception has concentrated on how and when the discrimination and categorization of speech sounds change with specific linguistic experience. It has become increasingly clear, however, that there are also universal biases in place early in development that guide and constrain how perceivers from diverse linguistic backgrounds decode the speech signal. In the domain of vowel perception, it is now known that perceivers (both adult and infant) are universally biased toward articulatorily and acoustically extreme vowels. This generic vowel bias is often demonstrated in discrimination tasks as a directional asymmetry: perceivers perform better when discriminating changes from less to more peripheral vowels compared to the reverse. I will discuss evidence indicating that the processes underlying these asymmetries operate on articulatory information, rather than on acoustic information *per se*. I will begin with findings from cross-language experiments with adults indicating that asymmetries occur with vowels presented in either the auditory or the visual modality, regardless of native language. I will then present findings indicating that analogous asymmetries in visual perception emerge using schematic non-speech visual analogs of vowels, but only if the optical stimuli depict both lip-motion and configural information, consistent with an articulatory account of asymmetries.

9:30

1aSC2. Speech perception frameworks revisited using the complex auditory brainstem response. Tian Zhao (Inst. for Learning and Brain Sci., Univ. of Washington, University of Washington, Box 367988, Seattle, WA 98195, zhaotc@uw.edu) and Patricia K. Kuhl (Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA)

Speech perception and experience-related effects have been theorized and experimentally tested through a wealth of behavioral measures as well as neural measurements focusing on cortical processes. Recently, the complex auditory brainstem response, targeting neural processes at a much earlier stage, has allowed us to generate and test new research questions from existing theoretical frameworks and can potentially expand these frameworks. We will present results from two recent studies examining different components of complex auditory brainstem responses to test hypotheses regarding universal as well as experiential effects in speech perception. In the first study (Zhao and Kuhl, 2018), the onset latency of the complex auditory brainstem response was found to predict perception of stop consonant and that both onset latency and perception are modulated by language background. In the second study (Zhao *et al.*, 2019), the frequency-following component of the complex auditory response (FFR) was found to show directional asymmetry for the neural discrimination of an English vs. French prototypical /u/, supporting the Natural Reference Vowel framework. Implications and future directions will be discussed regarding using complex auditory response to test speech perception frameworks.

9:55

1aSC3. Language-general and language-specific influences on Dutch and Japanese infants' perception of place of articulation. Sho Tsuji (Univ. of Tokyo, The University of Tokyo Int. Res. Ctr. for Neurointelligence (IRCN), 7-3-1 Hongo Bunkyo-ku, Tokyo 113-0033, Japan, tsujish@gmail.com)

Perceptual asymmetries provide a window into the role of language-general biases and language-specific experiences in early phonetic development. We assessed infant perception of two asymmetries related to coronal place of articulation and documented across the inventories of languages. To disentangle the role of language-general tendencies and experiential factors, we compared early perception of infants learning Dutch (where the input is consistent with language-general tendencies) and Japanese (where this is not the case). First, we document that both Dutch and Japanese infants start out with asymmetric discrimination of labial-coronal consonant contrasts

consistent with language-general predictions at 4–6 months of age, but that, consistent with input-based predictions, this pattern declines in Japanese, but not Dutch toddlers by 18 months of age. Second, we show that infants from both language groups develop phonotactic preferences consistent with their input but with different timelines, with French infants developing a labial-coronal preference by 10 months, and Japanese infants a delayed coronal-labial preference by 13 months of age. Together, these results demonstrate that language-specific input characteristics start overriding language-general tendencies, but also suggest that these language-general tendencies continue playing a role in perception. We speculate about sources of these biases on the acoustic, perceptual, and articulatory level.

10:20–10:35 Break

10:35

1aSC4. Perceptual biases in consonant perception—Where do we find them? What is behind them? Linda Polka (School of Commun. Sci. & Disord., McGill Univ., 2001 McGill College Ave., 8th Fl., Montreal, PQ H3A 1 G1, Canada, linda.polka@mcgill.ca)

There is no doubt that language experience has an early and profound impact on the perception of phonetic segments. There is now clear evidence that phonetic perception is also shaped by universal perceptual biases that can be exposed as directional asymmetries in phonetic discrimination. Much of this work has focused on vowel perception in which we observe robust perceptual asymmetries in infant and adult L2 perception. To explain these patterns, Polka and Bohn (2003) and (2011) outlined the Natural Referent Vowel (NRV) framework which proposes that vowel perception is shaped by both generic (universal) and language-specific processing. In this talk, I will present data showing perceptual biases in consonant perception which suggest that we can extend NRV principles to consonant perception.

11:00

1aSC5. Perceptual asymmetry in lexical tone perception. Ratree Wayland (Linguist, Univ. of Florida, 2801 SW 81st St., Gainesville, FL 32608, ratree@ufl.edu), Si Chen (Dept. of Chinese and Bilingual Studies, Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), Yitian Hong (Dept. of Chinese Lang. and Lit., Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), and Zhou Fang (Dept. of Chinese and Bilingual Studies, Hong Kong Polytechnic Univ., Hong Kong, Hong Kong)

Directional asymmetries have been documented in both infant and adult perception of lexical tones. For example, 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. Yeung *et al.* (2013) reported that 4- and 9-month-old Mandaring learning infants are better at Cantonese tone discrimination after being familiarized with T2 (25) than with T3 (33). Francis and Ciocca (2003) found that native Cantonese speakers' tone discrimination was better when the first syllable was higher in frequency (about 4 Hz) than the second syllable. Finally, in an ERP study, 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. In this talk, we will report results from two studies examining the effects of memory load and first language on perceptual asymmetry patterns in adult lexical tone perception among native English, Mandarin, and Cantonese speakers. Application of theoretical models including the perceptual magnet effects and perceptual assimilation will be discussed.

11:25

1aSC6. Experiential effects in speech perception: Do they arise from a level playing field? Catherine T. Best (MARCS Inst., Western Sydney Univ., Locked Bag 1797, Penrith, New South Wales 2751, Australia, c.best@westernsydney.edu.au)

Much evidence has accrued indicating that language experience from infancy shapes our categorization and discrimination of consonants and vowels, differentiating perception of native versus non-native speech contrasts. The Perceptual Assimilation (PAM: Best), Speech Learning (SLM: Flege), Native Language Magnet (NLM: Kuhl) and Second Language Linguistic Perception (L2LP: Escudero) models each offer theoretical accounts of this attunement but do not explicitly consider whether it begins with perceptual equivalence across all speech contrasts. Other findings suggest the initial perceptual playing field Is not level – some contrasts are intrinsically easier than others to discriminate. This paper considers two theoretical accounts of this uneven initial state, the Natural Referent Vowel framework (NRV: Polka and Bohn) and the Articulatory Organ Hypothesis (AOH: Goldstein), in light of additional findings of discrimination biases. We will discuss how initial biases may interact with language experience to modulate perceptual attunment to the native language.

Session 1aSP

Signal Processing in Acoustics, Architectural Acoustics, and Noise: Signal Processing for Architectural Acoustics and Noise Control I

Matthew S. Byrne, Cochair
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Kainam Thomas Wong, Cochair
Beihang University, School of General Eng., Beijing 100083, China

Chair's Introduction—9:45

Invited Papers

9:50

1aSP1. Characterization of the effects of ground boards on acoustic signals. Mark Anderson (Bringham Young Univ., MS 461, Hampton, VA 23681, anderson.mark.az@gmail.com), James H. Stephenson (Aviation and Missile Ctr., Aviation Development Directorate, US Army Combat Capabilities Development Ctr., Hampton, VA), Nikolas S. Zawodny (NASA Langley Res. Ctr., Hampton, VA), and Kent L. Gee (Bringham Young Univ., Provo, UT)

International and Federal regulations stipulate the acquisition of aircraft noise shall be conducted using inverted pressure microphones over a round ground board. These ground boards are used to provide an acoustically hard reflecting surface, limiting the effects of the potentially absorptive ground local to the test location. The microphone location is also specified to be offset from the center of the board to limit the effects of acoustic diffraction off the board edge. In order to determine the effects of the ground board on the measured acoustic signal, a comprehensive measurement campaign was undertaken at NASA Langley Research Center. The experimental setup included multiple ground board configurations placed on top of a sand pit, in an otherwise anechoic chamber. Ground board configurations included a microphone inverted and offset over the round ground board, a microphone offset and flush mounted in the ground board, and a microphone flush mounted in the sand. A detailed frequency sweep from a known source was then used to investigate the ground board effects on the recorded signal. Normal impedance measurements were also acquired to determine the reflection coefficients of the sand and ground boards. Experimental results are discussed, along with implications for future research and development.

10:10

1aSP2. Bayesian optimization of optically transparent multilayer micro-slit panel broadband sound absorbers. Michael Hoeft (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, hoeftm@rpi.edu), Ning Xiang, and Cameron J. Fackler (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Bayesian analysis is applied to the design and fabrication of multilayer micro-slit panel (MSP) absorbers to increase the manufacturing accuracy of each panel. An iterative inversion is applied to quantify fabrication inaccuracies. Re-manufacturing the panels with compensated MSP parameters improves the accuracy of measured sound absorption coefficients. This study builds on previous research where optically transparent, broadband sound absorption is made possible using a two-tiered Bayesian inference approach. The Bayesian design method automatically determines the most concise number of layers required to achieve a desired sound absorption, simultaneously yielding the MSP parameters for each layer of the resulting composite. This work aims to show that design of an optically transparent sound absorber with a broad frequency range of high absorption can be iteratively optimized using Bayesian inference with a multi-step fabrication process.

10:30

1aSP3. Spatial Fourier transform-based localized sound zone generation methods with loudspeaker arrays. Takuma Okamoto (National Inst. of Information and Communications Technol., 3-5, Hikaridai, Seika-cho, Soraku-gun, Kyoto 619-0289, Japan, okamoto@nict.go.jp)

This presentation provides spatial Fourier transform-based localized sound zone generation methods with multiple loudspeakers. First, these approaches are compared with conventional acoustic contrast and pressure matching approaches which are based on the least squares (LS) solution. Whereas these LS-based methods are unstable and some regularization schemes are required because the acoustic inverse problem is very ill-conditioned, the proposed spatial Fourier transform-based approaches can directly derive stable driving functions of loudspeakers without regularizations since the propagation and evanescent components of sound fields can be separated and only the propagation components are introduced to the driving functions. Then, spatial Fourier transform-based multiple sound zone generation methods with linear and circular loudspeaker arrays are introduced. In these approaches, the sound pressures on a line or a circle are modeled as rectangular or Hann windows, and the driving functions are analytically derived from the spatial Fourier transform. Additionally, localized sound zone generation approaches with linear, circular, and spherical loudspeaker arrays are introduced. In these approaches, the sound pressures produced by a loudspeaker are cancelled by the linear, circular, and spherical loudspeaker arrays. These driving functions are also derived from the spatial Fourier transform. Challenges and prospects of these approaches are finally described.

10:50

1aSP4. Continuous scan beamforming with CLEAN-SC for mapping of highly varying amplitude acoustic sources with significantly restricted sensor budgets. Abe H. Lee (ATA Eng., Inc., 13290 Evening Creek Dr. South, Ste. 250, San Diego, CA 92128, abe.lee@ata-e.com), Parthiv Shah, and Andrew White (ATA Eng., Inc., San Diego, CA)

Continuous scan beamforming (CSBF) is a novel approach that employs virtual sensors to improve array performance without increasing the physical number of sensors. Previous CSBF demonstrations (at the Spring 2019 ASA meeting in Louisville, KY) showed the capability to perform high resolution mapping of acoustic sources and identify almost an 18 dB difference in source level, which was not possible using conventional beamforming with an equal sensor budget. The current work investigated the capability of CSBF combined with CLEAN-SC for conducting source localization with significantly reduced sensor budgets (on the order of ten sensors), in which conventional beamforming is practically impossible to obtain appreciable results. Results of a four-speaker noise test showed that CSBF can perform accurate identification of the locations and levels of the sources with such restricted sensor counts, and can provide an accurate input to CLEAN-SC for further enhancement of the source map.

11:10

1aSP5. Measurement of the directional properties of non-diffuse sound fields in ordinary rooms. Mélanie Nolan (Saint-Gobain Ecophon, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, melnola@elektro.dtu.dk) and Erling Nilsson (Saint-Gobain Ecophon, Hyllinge, Sweden)

Reverberation time measurements are often inadequate in predicting the acoustical behavior of small ordinary rooms, particularly in spaces with non-uniform placement of absorptive materials (such as classrooms, offices, and clinical rooms). On the other hand, numerical predictions depend strongly on the quality of input data (i.e., absorption and scattering coefficients), which are generally inaccurate due to the lack of reliable experimental methods to measure them. This work investigates how spatially distributed measurements can be used to characterize the spatio-temporal properties of the sound field in non-Sabine spaces, and how this information can supplement traditional methods in tasks related to room acoustical design. In this study, measurements are conducted in a classroom with absorbing ceiling using a programmable robotic arm. It is shown that by expanding the measured sound field into an elementary wave basis, it is possible to reconstruct and extract the spatial distribution of sound pressure and energy flows in the room. Additionally, spatio-temporal post-processing enables to characterize the decay process and to assess the dominant directions of sound propagation, leading to a time-dependent analysis of the properties of the sound field.

11:30

1aSP6. Sparse representation of the sound field in a room with dictionary learning. Efren Fernandez-Grande (Acoust. Technol., Dept. of Elec. Eng., DTU - Tech. Univ. of Denmark, Ørsteds Plads, B. 352, DTU, Kgs. Lyngby DK-2800, Denmark, efg@elektro.dtu.dk), Manuel Hahmann, and Samuel A. Verburg (Acoust. Technol., Dept. of Elec. Eng., DTU - Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

Phased array measurements of the sound pressure in a room enable to reconstruct the sound field, i.e., to estimate pressure, velocity and sound intensity in positions that have not been measured. Typically, analytical wave functions are used to expand the measured data and interpolate the wave field. However, these bases are often redundant and lead to non-sparse solutions, as multiple basis functions are required to represent the measured data. In this study, we examine the use of dictionary learning to obtain a sparse representation of the sound field in a room, using atoms learned from experimental data. The aim is to obtain a model of reduced dimensionality that can represent optimally the spatial properties of the sound field in a room. We analyse the properties of the extracted dictionaries, their ability to reconstruct the sound field, and their generality. A broader question is the suitability of a given dictionary, which has been extracted from a particular room, to represent the sound field in another room.

Session 1aUW

Underwater Acoustics: Underwater Acoustic Data Communication

Zhiqiang Liu, Cochair
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Aijun Song, Cochair
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Contributed Papers

9:00

1aUW1. Preliminary estimates of year-round acoustic communications potential in the Canada Basin. Carolyn Binder (Defence R&D Canada, LSC Ocean Wing, 1355 Oxford St., PO Box 15000, Halifax, NS B3H 4R2, Canada, carolyn.binder@dal.ca), Sean Pecknold (Defence R&D Canada, Dartmouth, NS, Canada), and Mohsen Badiey (Univ. of Delaware, Newark, DE)

Long-range acoustic communication networks, which operate over hundreds of kilometers, are important enablers for persistent surveillance in the Arctic. Acoustic communication must take place over such long ranges due to difficulty in deploying communication nodes in such an inhospitable environment. Signal transmission over long ranges is challenged by distortion caused by refraction and multipath addition, so the propagation medium should be understood in order to optimize node placement and estimate the network performance. The Arctic is a unique acoustic propagation environment due to the presence of an ice-covered surface and oceanographic variability around the shelf breaks. To better understand the current state of Arctic propagation, acoustic signals were transmitted during the year-long Canada Basin Acoustic Propagation Experiment (CANAPE). In this talk, signals from the CANAPE data are used to assess the feasibility of year-round, long-range acoustic communications in the Canada Basin by examining several metrics. These metrics include: the signal-to-noise ratio (SNR) of the received signals, time-spreading of the signals due to multipath arrivals, and signal coherence. This talk will focus on the seasonality of these metrics in order to assess how the performance of acoustic communication systems may be altered by temporal variability of the Arctic acoustic propagation characteristics.

9:15

1aUW2. Experimental investigation of acoustic channel reciprocity in shallow water. Zhiqiang Liu (US Naval Res. Lab., 4555 Overlook Ave., Washington, DC 20375, zhiqiang@ieee.org), Lloyd Emokpae, Jeffrey Schindall, and Geoffrey F. Edelmann (US Naval Res. Lab., Washington, DC)

Channel reciprocity is recognized as one of the most fundamental properties of underwater acoustic propagation. The theoretical proof of channel reciprocity assumes that the ocean environment is static during the course of acoustic propagation. Since this assumption might not be valid for the case of shallow-water acoustic propagation, a question to be raised is: does channel reciprocity remain true in shallow water? A sea-going experiment was recently conducted to study the shallow-water channel reciprocity. In the experiment, two custom-built software-defined acoustic modems were used to probe their shared channel from two opposite directions. The channel reciprocity was examined by comparing the two probed channels directly. The probing signals and handshaking protocols were judiciously designed to minimize the effects of channel dynamics under the constraint of half-duplex acoustic modems. This paper will provide the details of this sea experiment (including its design, setups and execution) and report interesting findings. [Work supported by US Office of Naval Research.]

9:30

1aUW3. Impact of channel fluctuations on channel estimation errors in underwater acoustic communications. Zheng Guo (Elec. Engi., Univ. of Alabama, Tuscaloosa, AL) and Aijun Song (Elec. Eng., Univ. of Alabama, 401 7th Ave., Hardaway Rm. 284, Tuscaloosa, AL 35487, song@eng.ua.edu)

Channel estimation is critical to achieve high data rate acoustic communications in the ocean. Channel estimates are often utilized to address the distortions induced by multipath propagation in various communication receivers. Therefore, accurate channel estimation is often the prerequisite for reliable coherent acoustic communications. Many efforts have been devoted to either characterizing the acoustic channel or developing high-performance channel estimation algorithms. However, limited work has been directed to investigate effect of channel fluctuations on estimation performance. Here we seek to quantify the impact of channel fluctuations on least squares channel estimators. A new metric, channel variation ratio, is used to describe the rate of fluctuations in the acoustic impulse responses. We investigate the relationship between the mean squared error (MSE) of the channel estimates and the channel variation ratio. We show the new metric can be used to predict channel estimation MSE for least squares channel estimators. Numeric results also show that there exists an optimal channel length with the minimum estimation error for fluctuating acoustic channels. Both computer simulations and experimental data have been used to validate the findings. [Work supported by the National Science Foundation CNS# 1704076.]

9:45

1aUW4. Iterative multi-channel FH-MFSK reception in underwater acoustic communication. Dajun Sun (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China), Xiaoping Hong (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Nangang Dist, Harbin, Heilongjiang 150001, China, xiaop5140@hrbeu.edu.cn), Hongyu Cui, and Lu Liu (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China)

With the challenges of long multipath, large Doppler spreads, high ambient noise and rapid phase fluctuation in mobile shallow underwater acoustic (UWA) channels, design of robust UWA communication system should be carefully considered. Aiming at this topic, a convolutionally encoded noncoherent frequency-hopped M-ary frequency-shift-keyed (FH-MFSK) modulation scheme with a novel iterative multi-channel reception is

proposed. To obtain the reception suitable for noncoherent MFSK, the ideas of iterative multi-channel demodulation and decoding are combined and extended. A modified multi-channel soft input soft output (SISO) demodulator based on maximum a posterior (MAP) criterion is derived, and substantial gains of several decibels in power efficiency are achieved. Extra gain can be obtained by utilizing time diversity with long multi-path delay. Simulation shows that a noncoherent receiver equipped with 5 arrays obtains around 5.5 dB gain over single noniterative receiver. A shallow water field testing at Songhua Lake with 500 m distance verifies the robustness and usability of the proposed receiver.

10:00

1aUW5. A channel state information updating method based on the first-order Gauss Markov model under time-varying underwater acoustic channel. Gang Qiao (Harbin Eng. Univ., Harbin, China), Lei Liu (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong Str, Nangang Dist, Harbin 150001, China, heu_liulei@hotmail.com), and Lu Ma (Harbin Eng. Univ., Harbin, China)

Because of the time-varying characteristic and the large propagation delay in underwater acoustic (UWA) channel, adaptive resource allocation in UWA orthogonal frequency-division multiplexing access (OFDMA) system cannot be performed with the assumption of perfect channel state information (CSI) at the transmitter. Therefore, this paper focuses on the imperfect CSI and proposes a CSI updating method to reduce the impact of imperfect CSI on the performance of resource allocation under time-varying UWA channel. First, we analyze the channel estimation error of OFDM system under time-varying UWA channel and define the per-subcarrier channel temporal correlation (PSCTC). Finally, a CSI updating method based on the first-order Gauss-Markov model (GM-CSI) is proposed for optimizing CSI feedback in the UWA channel with large propagation delay, where the channel estimation error and PSCTC factor are considered as two influencing factors in this model. Simulation results show that the GM-CSI can effectively mitigate the effect of time on CSI effectiveness, and the resource allocation by using GM-CSI has higher effective throughput and lower BER than that of imperfect CSI.

10:15–10:30 Break

10:30

1aUW6. Underwater acoustic communication performance by signal to noise plus interference ratio in BLAC18. Hyeonsu Kim (Marine Sci. and Convergence Eng., Hanyang Univ., 55, Hanyangdaehak-ro, Sangnok-gu, Ansan, Gyeonggi-do 15588, South Korea, hskim00@hanyang.ac.kr), Jee Woong Choi (Marine Sci. and Convergence Eng., Hanyang Univ., Ansan, South Korea), and Ho Seuk Bae (Agency for Defense Development, Changwon-si, South Korea)

In underwater acoustic communications, inter-symbol interference (ISI) caused by multipath propagation of underwater channel degrades communication performance significantly. Various techniques, such as time reversal combining and decision feedback equalization, have been developed to eliminate the influence of the interference. However, few studies have assessed the performance with the signal-to-noise (SNR) of received communication signals. In practical underwater acoustic systems, the channel must be accurately estimated for the channel equalization and the SNR is involved in the accuracy of the channel estimation. The effect of interference is dominant at short distances, but the effect of noise becomes dominant as distance increases. In this study, SNR, signal-to-interference ratio, and signal-to-noise plus interference ratio (SINR) are estimated as a function of the distance, and communication performances are examined. Communication data from BLAC18, conducted in East Sea of Korea in 2018, are used to analyze the communication performances. In the experiment, communication signals with a carrier frequency of 2.5 kHz were transmitted and received at horizontal distances of 30, 60, and 90 km. Data analysis results shows that communication performance is most correlated with SINR. [Work supported by the ADD(UD170022DD).]

10:45

1aUW7. Vector communication by moving source during KOREX-17. Kang-Hoon Choi (Marine Sci. and Convergence Eng., Hanyang Univ., Hanyangdaehak-ro 55, Sa 1-dong, Sangnok-gu, Ansan-si, Gyeonggi-do 15588, South Korea, choikh0210@hanyang.ac.kr), Hyeonsu Kim (Marine Sci. and Convergence Eng., Hanyang Univ.,Ansan, Gyeonggi-do, South Korea), Jee Woong Choi (Marine Sci. and Convergence Eng., Hanyang Univ.,Ansan, South Korea), Sunhyo Kim (Maritime Security Res. Ctr., Korea Inst. of Ocean Sci. and Technol., Ansan, South Korea), and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

In underwater acoustic communication, a time reversal technique with spatially separated large-size receiver array has been used to overcome delay spread caused by multipath channel and obtain spatial diversity gain. This technique is difficult to be accommodated in space-constraint environments due to the size of receiver array. Since an acoustic vector sensor has capability to simultaneously measure x, y, and z-components of particle velocities as well as acoustic pressure at a single point, it can provide better communication performances than a system using hydrophones only. In this study, acoustic communication performance using a vector sensor is demonstrated using communication data collected during Korea Reverberation Experiment (KOREX-17) conducted in shallow water located at 34°43′N, 128°39′E on May 25, 2017. The channel characteristics and communication performances extracted using the acoustic pressure and particle velocity components are shown in this talk, and the performance variation is discussed as a function of distance. [Work supported by the Ministry of Oceans and Fisheries, Korea (PMS4110).]

11:00

1aUW8. Application of ray-based blind deconvolution to long-range acoustic communications. Donghyeon Kim (Ocean Sci. and Technol., Korea Maritime and Ocean Univ., 727 Taejong-ro, Yeongdo-Gu, 253, Busan, South Korea, donghyeonkim@kmou.ac.kr), Heejin Park, Jea Soo Kim (Korea Maritime and Ocean Univ., Busan, South Korea), and Joung-Soo Park (Agency for Defense Development, Changwon, Gyeongnam, South Korea)

For unknown source waveforms, the channel impulse response (CIR) can be estimated by ray-based blind deconvolution (RBD). In previous papers, RBD was successfully demonstrated using the simulation and data in a shallow water environment. In this study, we investigate the applicability of RBD for a long range (e.g., 30 km, 60 km, and 90 km) in about 1000 m deep water, using experimental data recently conducted in the east of Pohang, South Korea, in October 2018. The goal of BLAC18 was to obtain the underwater communication data in deep water. Simulation and data results are presented to demonstrate CIR estimation of a communication signal (2.2–2.9 kHz) using a 16-element, 42-m long vertical array in approximately 1000 m deep water. The results show that the CIR estimated from RBD is comparable to that of matched-filter result. Additional communication performance comparison will be also presented.

11:15

1aUW9. Low complexity energy efficient orthogonal frequency division multiplexing communication system over underwater acoustic channel by partial transmit sequence peak to average power ratio reduction. Waleed Raza (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145 Nantong St., Harbin, Heilongjiang 150001, China, waleedraza93@gmail.com), Xuefei Ma, Tingting Wang, and Muhammad Bilal (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin 150001, China)

We discourse the power consumption for underwater acoustic transceivers in this study. We have proposed a less computational complex underwater acoustic OFDM communication system. Underwater acoustic transceivers consume more power than that Radio frequency transceivers. The methods which are being utilized in Radio frequency cannot directly be applied to underwater acoustic system. Therefore, it needs to re-investigated and the new techniques should be introduced to achieve reliable and energy

efficient acoustic data transmission. It also relies upon condition of acoustic channel and transmission distances. There is long-term battery-operated deployment of underwater acoustic modems in some applications. To maintain sustainable network operations an energy efficient OFDM system is required. In this paper, we have used partial transmit sequence peak to average power ratio reduction technique to design a novel energy efficient OFDM system for underwater acoustic channel with less complexity. This framework will be remarkably helpful in future for long term battery deployed applications of underwater acoustic transceivers. The results are expected to provide the enhanced performance of PAPR reduction with better sustainability. It has less complexity and better energy efficiency for underwater acoustic sensor/sensing networks.

MONDAY AFTERNOON, 2 DECEMBER 2019 WINDSOR, 1:00 P.M. TO 5:00 P.M.

Session 1pAA

Architectural Acoustics, Noise Structural Acoustics and Vibration, and ASA Committee on Standards: Sound Transmission and Impact Noise in Buildings

Matthew V. Golden, Cochair
Pliteq, 4211 Yonge St., North York, Ontario M2P 2A9, Canada

Benjamin Shafer, Cochair
Technical Services, PABCO Gypsum, 3905 N 10th St., Tacoma, Washington 98406

Chair's Introduction—1:00

Invited Papers

1:05

1pAA1. Hard surface flooring policy for residential condominiums. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

A majority of the residential condominiums in the USA were originally constructed with wall to wall carpet flooring in the living rooms, bedrooms, and hallways. Removing the carpet and replacing it with various types of hard surface flooring is a recent popular trend. Converting these areas from carpet to hard surface flooring will result in a significant increase in impact sound transmission to the floor below as well as adjacent living units on the same floor. Unfortunately, some condominium homeowner's associations do not have a written policy for accepting these modifications. Those that do often set the floor acceptance criterion as a field tested FIIC (now called AIIC) rating (typically in the 50 to 55 range), tested after the hard surface flooring is installed. This can result in significant financial impacts to the homeowner installing the flooring if final testing reveals that the floor does not meet the criterion set by the homeowner's association. This presentation will discuss the complicated issues facing all three parties (homeowner, neighbor, and HOA Board) and presents a recommendation for a written policy that is fair to all.

1:25

1pAA2. A new rating method for low-frequency impact noise. Ryan L. Skoug (ESI Eng., 7831 Glenroy Rd., Ste. 430, Minneapolis, MN 55439, rskoug@esi-engineering.com)

Residents, building owners, architects and acoustic consultants are rightly concerned about footfall noise transmission through multi-family building floor/ceiling assemblies. By code, most projects in the USA are required to meet a minimum design rating of IIC 50; AIIC 45 if field tested. However, no matter how high the rating, residents in wood frame buildings often complain that low-frequency, or "thudding" footfall noise is a problem. Unfortunately, the IIC rating system does not evaluate impact noise at frequencies below 100 Hz. In this paper, we show that footfall noise is present at audible frequencies below 100 Hz, how this footfall noise can be replicated more objectively using ball drops or a tapping machine, and how measured impact noise might be used to develop a single number rating for low-frequency impact noise insulation of field tested floor/ceiling assemblies. The findings presented are not meant to be taken as final results, but rather a summary of progress made to-date for work that is under-development.

1:45

1pAA3. Measurement of low frequency impact insulation. David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, wdong@veneklasen.com) and John LoVerde (Veneklasen Assoc., Santa Monica, CA)

The authors have proposed a dual-rating method for measuring impact insulation in which the high and low-frequency components of impact noise are evaluated independently [LoVerde and Dong, *J. Acoust. Soc. Am.* **141**, 428–440 (2017)]. The measurement of low frequency impact sound in rooms is critical to determining human response and acceptability of floor ceilings. In this paper we discuss issues related to low frequency measurements in residential applications. The paper will include analysis of the suitability of the standard tapping machine as a source for low frequency impact noise for typical assemblies within North America. The authors will review the uncertainty of low frequency measurements in typical receiving rooms. The paper will discuss the measurement methodology as a requirement for adoption of a measurement standard.

2:05

1pAA4. Effect of underlayment formulation and structural design on high and low frequency impact isolation metrics. Matthew V. Golden (Pliteq, 4211 Yonge St., North York, ON M2P 2A9, Canada, mgolden@pliteq.com) and Wilson Byrick (Pliteq, Toronto, ON, Canada)

The ASTM E33 Building acoustics committee is currently evaluating two proposed new standards on high and low frequency impact noise ratings. Unlike the Impact Isolation Class (IIC) which covers a frequency range of 100 to 3150 Hz, the new high and low frequency ratings cover ranges of 50–80 Hz and 400–3150 Hz, respectively. Correlations among these new ratings and the standard IIC rating for several hundred laboratory tests will be show along with examples of their usefulness. A detailed analysis of several different structural assemblies will be shown. Finally, effects of underlayment formulation on the new ratings and IIC are discussed.

2:25

1pAA5. The problem with E90. Corey Taylor (Owens Corning, 2790 Columbus Rd., B75, Granville, OH 43023, Corey.taylor@owenscorning.com) and Kevin Herreman (Owens Corning, Granville, OH)

Wall constructions are typically evaluated at an accredited acoustical laboratory for sound transmission loss per the ASTM E90 standard and the STC determined per the ASTM E413 standard. In North America, there are several accredited acoustic laboratories where walls can be evaluated. Accredited acoustic laboratories are required by the ASTM E90 standard to demonstrate that their facility is in compliance with the standard by performing transmission loss testing utilizing a reference specimen outlined in the ASTM E1289 standard. This standard prescribes a series of reference assemblies based on sheet metal panels in a metal frame, all with STC values less than 30, to be used in demonstrating compliance. A recent study conducted at the Owens Corning Acoustic Research Center, an accredited acoustical laboratory, provided enough data to perform a gauge study on the E90 test. The results of that study will be presented in this paper.

2:45

1pAA6. Laboratory sound transmission loss testing for steel-framed partitions II: Stud spacing and steel material properties. Benjamin Shafer (Tech. Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406, ben.shafer@quietrock.com)

There exists a large variation in laboratory sound transmission loss (STL) results and calculated STL-based ratings. Multiple research studies have found that this laboratory variation is consistent over multiple test series and assemblies, but more especially for steel-framed assembly designs. A previous research study presented at the Acoustical Society of America illustrated and quantified the STL difference for various steel mil thicknesses from 15-mil nonstructural to 97-mil structural. This research study is the second phase of this steel-framed STL research study and it investigates stud spacing and steel material- and shape-characteristics. The effect that each of these properties has on the STL of steel-framed partitions will be illustrated and discussed.

3:05–3:20 Break

3:20

1pAA7. Weight loss is good, right? Kevin Herreman (Owens Corning, 2790 Columbus Rd., B75, Granville, OH 43023, kevin.herreman@owenscorning.com) and Corey Taylor (Owens Corning, Granville, OH)

Since the introduction of lightweight drywall there has been concern related to the effect on the sound transmission loss of wall systems built using these products due to the reduced weight. Maintaining drywall strength while reducing weight 20 to 30 percent should improve installation productivity and worker safety on the build site. The process of reducing the weight is likely accomplished by entraining more air pockets into the slurry during manufacturing thus displacing a portion of the gypsum in the board. However, those familiar with the performance of wall systems know that the mass of the wall contributes to the overall sound transmission performance. A reduction in mass, of which the drywall is a significant portion, should reduce the transmission loss of the wall. This paper will highlight test data and modeling from the Owens Corning Acoustic Research Center demonstrating how these losses could be overcome.

Contributed Papers

3:40

1pAA8. In-line syntactic-foam device for control of water hammer and fluid-borne noise. Kenneth Cunefare (Georgia Tech, Mech. Eng., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu), David N. Ramsey, and Nathaniel Pedigo (Georgia Tech, Atlanta, GA)

Water hammer, a common problem in plumbing, is characterized by an impulsive increase in pressure. The pressure rise and associated momentum transfer may cause plumbing noise and vibration, and, potentially, catastrophic plumbing component failures. Water hammer arresters are required in building codes to be proximate to fast-acting valves, such as present in washing machines and other appliances. Commercially available water hammer arresters introduce compliance with a free-piston air spring as the means to limit the peak pressure during water hammer. Syntactic foam, comprising microspheres within a host polymer matrix, may be suitable for application in a water hammer arrester device. Foam-based arresters also enable flow-through designs, which may reduce other fluid-borne noise in addition to controlling water hammer. A prototype flow-through, in-line syntactic water hammer arrester device has been designed and tested. The flow-through design demonstrates adequate performance for service as a water hammer arrester, as well as significant insertion loss against fluid-borne noise.

3:55

1pAA9. Designing the quietest room in the world. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

In 2015, the Guinness Book of World Records identified the anechoic chamber in Microsoft's Building 87 in Redmond, WA as the quietest room in the world with an overall A-weighted background noise level of -21 dB. This presentation discusses the original design criteria for the room, the process of evaluating the acoustic characteristics of the proposed site, and steps that were taken to achieve the ultimate goal. The instrumentation that was used to measure the background noise level will also be discussed.

Invited Paper

4:10

1pAA10. Massive or lightweight? Sound insulation of Austrian multistory apartment buildings since 2000. Heinz Ferk (Bldg. Sci. Lab, Graz Univ. of Technol., Inffeldgasse 24, Graz, Styria 8010, Austria, ferk@tugraz.at)

During the past centuries, Austria was a country of massive multistory apartment houses. Since the last decade of the last century, technical and legal developments led to experimental multistory buildings with lightweight construction as timber frame or cross laminated timber construction. During the following years, the proportion of lightweight constructions increased slightly. In Styria, a federal land in the south of Austria, it is usual to perform sample performance measurements also of the quality of subsidized apartment buildings upon competition. This paper shows some results of such quality measurements of airborne and impact sound insulation during the past two decades and discusses the performance of different constructions. Moreover, some results of "mikrozensus" surveys about noise pollution of the federal institution "Statistik Austria" are presented. The question arises, if the used descriptors as, e.g., the weighted sound reduction index R_w are still sufficient meaningful in every case or if there should be some additional information to avoid specific problematic construction solutions. Another way would be to provide rules for robust in detail. Some special cases and construction influences are shown exemplarily. For future development, it would be of interest to provide more information about sensitivity of construction measures in order to enhance the future sound insulation performance not only as a number but in reality.

Contributed Papers

4:30

1pAA11. Sound transmission analysis of various lightweight wall panels. Nishant Kumar (Acoust. and Vib. Metrology and Elec. and Instrumentation Eng. Dept., CSIR-National Physical Lab., New Delhi and Thapar Inst. of Eng. and Technol., Patiala, Acoust. and Vib. Metrology, Main Bldg., Dr KS Krishnan Marg, Pusa, New Delhi, Delhi 110012, India, kumarnishant.kumar9@gmail.com) and Mahavir Singh (Acoust. and Vib. Metrology, CSIR-National Physical Lab., New Delhi, Delhi, India)

Demands of lightweight wall panels and acoustical privacy in recent years have accelerated the incorporation of wall panels in the modern building element constructions. Lightweight wall panels are based on multi-layered structures which are a combination of gypsum board, glass wool/mineral wool, resilient steel channels and staggered wood studs used for inter-dwelling walls in modern buildings. The objective of this paper is to investigate the implementation of different types of combination to improve the sound insulation of sandwich constructions from above materials. The Sound Transmission Class (STC) characteristics of panel constructions with various combinations have been investigated. The measurements have been performed in the sound transmission suite of the CSIR-National Physical Laboratory, India according to ISO 10140 and ASTM 90. This paper presents the STC value of 20 samples tested in systematic evaluation for sound transmission through different sandwich structures with a frequency range of 50–6300 Hz with STC varying 46–59. As low frequency sound is the common cause for noise, the sound transmission measurements have been done on frequencies as low as 50 Hz. This work will help the noise control engineers, builders, designers, acoustic consultants, and residents/consumers to select construction suitable for wall panels to use in buildings.

4:45

1pAA12. Introduction in the Russian federation of international standards in the field of measurement and evaluation of sound insulation. Ilya E. Tsukernikov (Acoust. Lab., Res. Inst. of Bldg. Phys., Odoevskogo proezd, h.7, korp.2, Fl. 179, 21 Lokomotovny pr., Moscow 117574, Russian Federation, 3342488@mail.ru), Igor Shubin (Res. Inst. of Bldg. Phys., Moscow, Russian Federation), and Tatiana Nevenchannaya (Technikal Mech., Moscow Politechnik Univ., Moscow, Russian Federation)

The features of the introduction in the Russian Federation of international standards for measurement and evaluation of sound insulation are considered. Over the past eight years, a system for measuring and evaluating the sound insulation of buildings and their elements has been created on the basis of international standards. In 2012, a series of international standards ISO 10140 for methods of laboratory measurements of airborne and impact

sound insulation was introduced as the national standards of Russia GOST R ISO. In 2016, the standards ISO 717 were introduced as modified national standards GOST R 56769-2015 and GOST R 56770-2015, setting the rules for single-number rating of airborne and impact sound insulation. In 2018, the international standard ISO 12999-1 was introduced as a modified national standard GOST R 57900-2017 for determining and applying the uncertainties in measuring sound insulation. In 2019, a draft national standard was prepared for field measurement of sound insulation of facades and their elements, which is a modified edition of the ISO 16283-3. Single-numeric parameters of airborne and impact insulation determined using these standards are subject to comparison with the normative values regulated by the national regulatory document—the set of rules SP 51.13330.

MONDAY AFTERNOON, 2 DECEMBER 2019 EDISON, 1:30 P.M. TO 5:15 P.M.

Session 1pAB

Animal Bioacoustics: General Topics in Animal Bioacoustics II

Colleen Reichmuth, Chair
Institute of Marine Sciences, Long Marine Laboratory, University of California, 1, 100 Shaffer Rd., Santa Cruz, California 95060

Contributed Papers

1:30

1pAB1. In-air hearing in Hawaiian monk seals: Implications for understanding the auditory biology of monachid seals. Brandi Ruscher-Hill (Dept. of Ocean Sci., Univ. of California Santa Cruz, 115 McAllister Way, Santa Cruz, CA 95060, bruscher@ucsc.edu), Jillian Sills, and Colleen Reichmuth (Inst. of Marine Sci., Univ. of California Santa Cruz, Santa Cruz, CA)

Despite its critically endangered status, little is known about the sensory biology of the Hawaiian monk seal (*Neomonachus schauinslandi*), a member of the Monachinae subfamily of phocid seals. While previous research has suggested that at least one monachid species exhibits reduced sensitivity to airborne sounds, no comparable hearing data are available for other species. Our aim in this study was to measure an in-air auditory sensitivity profile for a Hawaiian monk seal using a psychophysical paradigm. One adult male seal was trained to participate in a go/no-go procedure in an acoustic chamber, which enabled measurement of absolute detection thresholds for narrowband signals at frequencies spanning the range of hearing in air. The resulting behavioral audiogram for this individual reveals poor hearing sensitivity. The lowest threshold of 40 dB re 20 μPa at 0.8 kHz occurs within a +20-bandwidth of best hearing extending more than seven octaves. The apparently elevated in-air hearing thresholds for two monachid seals—compared to the highly sensitive true seals in the Phocinae subfamily—suggest that auditory adaptations may differ between these groups. Additional research is needed to confirm this finding, which has implications for understanding the evolution of hearing in amphibious marine mammals.

1:45

1pAB2. The genesis of giants: Behavioral, social, and vocal development of northern elephant seals. Caroline B. Casey (Inst. of Marine Sci., Univ. of California Santa Cruz, 100 Shaffer Rd., Santa Cruz, CA 95060, cbcasey@ucsc.edu), Isabelle Charrier (Equipe Communications Acoustiques, Neuro-PSI, Université Paris Sud, Orsay, France), Nicolas Mathevon (Equipe Neuro-Ethologie Sensorielle, ENES/Neuro-PSI, Univ. de Lyon/Saint-Etienne, Saint-Etienne, France), Claire Nasr (Dept. of Wildlife, Humbolt State Univ., Humbolt, CA), Parker Forman (Dept. of Vertebrate Ecology, California State Univ. Monterey Bay, Moss Landing, CA), and Colleen Reichmuth (Inst. of Marine Sci., Univ. of California Santa Cruz, Santa Cruz, CA)

To identify the conditions that support the development of hierarchical relationships within networks of familiar competitive rivals, we consider the ontogeny of area use, social relationships, and communicative behavior among male northern elephant seals (*Mirounga angustirostris*). We cross-sectionally sampled the behavior of 207 males during reproductive development on the breeding colony at Año Nuevo, CA to determine (1) whether space utilization changes as a function of age, (2) how social relationships vary across different age classes, and (3) the structural development of their ritualized acoustic displays. As males mature, space use within the breeding colony decreases, creating a more predictable social environment in which repeated interactions promote greater connectivity between individuals. Additionally, the vocalizations of sub-adult males transition from highly variable and unstructured to stable adult calls that reliably convey individual identity. The emergence of these acoustic signatures during a profound period of physical development—along with concurrent changes in movement patterns—coincides with the establishment of stable relationships between mature competitors. These findings indicate that ontogenetic changes in male behavior can be important determinants of adult position within structured hierarchies and provide insight into the role of behavioral development in species that exhibit intense competition for access to resources.

2:00

1pAB3. Vocal behavior in spotted seals (*Phoca largha*). Jillian Sills (Inst. of Marine Sci., Univ. of California Santa Cruz, 115 McAllister Way, Santa Cruz, CA 95060, jmsills@ucsc.edu) and Colleen Reichmuth (Inst. of Marine Sci., Univ. of California Santa Cruz, Santa Cruz, CA)

Captive studies can inform passive acoustic monitoring efforts by describing fundamental features of species-typical vocalizations emitted by known individuals. These include acoustic parameters as well as developmental, seasonal, and sex-based patterns in vocal behavior. Here, two male spotted seals were studied in captivity from age 3 months through 8 years. Vocal behavior was scored daily and opportunistically recorded. The production of underwater calls emerged with presumed sexual maturity (age 4). To evaluate vocal repertoire and fine-scale temporal patterns of sound production in adult seals, an underwater acoustic recorder was continuously deployed with these seals at age 7–8 years. The spotted seals produced a variety of underwater calls—including roars, knocks, and moans—with dominant energy below 1 kHz. There was a marked annual peak in vocal activity in spring, prior to the yearly molt. This period coincided with increased aggressive behavior and musky odor indicative of heightened reproductive activity. These results from developing male spotted seals, obtained in the absence of conspecific females, confirm the production of recognizable, stereotypic underwater calls associated with the breeding season. These findings can be used to inform the use of autonomous acoustic recorders to track the presence and movements of free-ranging seals in remote habitats.

2:15

1pAB4. Strain differences in song and hearing in canaries (*Sernius canarius*). Robert Dooling (Psych., Univ of Maryland, Baltimore Ave., College Park, MD 20742, rdooling@umd.edu), Jane brown, Beth Brittan-Powell, Gregory F. Ball (Psych., Univ of Maryland, College Park, MD), Matt Conte, Karen Carleton (Biology, Univ of Maryland, College Park, MD), and Farrah Madison (Psych., Univ of Maryland, College Park, MD)

Breeders have bred canaries either for specific song characteristics (song canaries) or morphology/plumage (type canaries) for centuries. Type canaries (e.g., Border and Gloster strains) retain song characteristics that are typically quite similar to those of wild canaries. By contrast, song canaries (e.g., Belgian Waterslager and Roller strains) have been selected for song types pleasing to the human ear, resulting in songs that, in most cases, are less complex, lower pitched, and narrower in a frequency range than songs from wild canaries. We now suspect that song selection in the Belgian Waterslager song canary has either directly or indirectly resulted in high-frequency hearing loss associated with hair cell abnormalities. Here, we compare hearing in the Belgian Waterslager and several other type and song canaries including the American Singer Canary. Though bred only since the 1930s, American Singer canaries also have a high-frequency hearing loss that looks very similar to that of the Belgian Waterslager and may have similar pathologies. Illumina whole genome sequencing has preliminarily identified a number of high-impact SnpEff variants in Belgian Waterslager and American Singer Canaries, some of which are related to deafness genes in mammals.

2:30

1pAB5. Propagation distances and sound properties of the antennal rasps produced by spiny lobsters (*Palinurus elephas*) in European coastal waters. Youenn Jézéquel (LEMAR, Laboratoire des Sci. de l'Environnement Marin (LEMAR), UMR 6539 CNRS, UBO, IRD, Ifremer, LIA BeBEST, Institut Universitaire Européen de la Mer (IUEM), rue Dumont D'Urville,12 Rte. de penhuel, Plouzané 29280, France, youenn.jezequel@univ-brest.fr), Julien Bonnel (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanographic Inst., Woods Hole, MA), Jennifer Coston-Guarini, and Laurent Chauvaud (LEMAR, Laboratoire des Sci. de l'Environnement Marin (LEMAR), UMR 6539 CNRS, UBO, IRD, Ifremer, LIA BeBEST, Institut Universitaire Européen de la Mer (IUEM), Plouzané, France)

Spiny lobsters (*Palinurus elephas*) have been overfished in European waters, and adult breeders are now scarce. Our recent study highlighted the high acoustic potential of this species, which can emit loud broadband pulse trains, called "antennal rasps," with peak-to-peak source levels (estimated at 1 m from the source) above 160 dB re 1 μPa^2 [Jézéquel *et al.*, *Marine Ecology Progress Series 615* (2019)]. These acoustic properties imply that these sounds could be detected during *in situ* passive acoustic monitoring. However, before using a such tool, we need to understand how antennal rasps propagate *in situ* and at what distance they could be detected above the ambient noise. To answer these questions, we recorded spiny lobster antennal rasps in the Iroise Sea (Brittany, France). We used a linear array of 8 hydrophones, with distances between animals and receivers ranging from 0.5 m to 100 m. We recorded antennal rasps from 38 individuals of various sizes. Our results demonstrate that large spiny lobsters can be detected at 100 m, and that sound properties might be directly influenced by the size of the individuals.

2:45

1pAB6. Alpha male Guyanan red howler monkey responses to nocturnal and diurnal loud calls. Leandro A. Do Nascimento (Dept. of Wildland Resources and Ecology Ctr., Utah State Univ., Logan, UT 84322, le_nasc@hotmail.com) and Karen Beard (Wildland Resources and Ecology Ctr., Utah State Univ., Logan, UT)

Our understanding of the communication system of howler monkeys is hampered by our ability to collect acoustic and behavioral data across their entire period of activity. It is not surprising that most studies are restricted to the daylight period despite knowledge that this species is also active at night. Our goal was to describe vocalization patterns, call structure, and test the function of diurnal and nocturnal loud calls of the Guyanan red howler monkey. To be able to study 24-h cycle call emissions, we used 20 passive acoustic monitoring devices deployed in the territory of four different troops of howlers. To test the function of diurnal and nocturnal loud calls, we set up a broadcast experiment and recorded the behavioral response of alpha males to the playbacks. There is a significant difference between some spectral properties of nocturnal and diurnal howls (duration, standard deviation, median frequency, skewness, kurtosis, and flatness). The alpha males responded differently to nocturnal and diurnal playbacks. They approached more often, vocalized more, and never evaded when presented with nocturnal call playbacks. We found significant differences between approach latency, evade latency, and soft call latency. Howler monkeys responded differently to the playbacks, which suggest that nocturnal and diurnal loud calls serve different functions. We found that nocturnal calls triggered more aggressive responses than diurnal calls.

3:00

1pAB7. Individuality in the vocalizations of adult and infant coppery titi monkeys (*Plecturocebus cupreus*. Allison Lau (Univ. of California, Davis, Young Hall, 1 Shields Ave., Davis, CA 95616, alljones@ucdavis.edu), Dena J. Clink (BioAcoust. Res. Program, Cornell Univ., Ithaca, OR), and Karen L. Bales (Univ. of California, Davis, Davis, CA)

Many primates use acoustic communication to maintain social relationships. Specifically, territorial, pair-bonding primates participate in coordinated vocal duets. Recognition of neighbors should be selected for, as identifying conspecifics may decrease the need for costly territorial behaviors. While similar species show vocal individuality, it is unknown if vocal individuality is innate or develops over time. To understand individuality across life stages, we analyzed 169 duet vocalizations from 30 adult titi monkeys (*Plecturocebus cupreus*) and 600 trills from 30 infants. We estimated 16 features of adult pulse-chirp vocalizations and 9 features of infant trills from spectrograms and used discriminant function analysis with leave-one-out cross-validation to classify individuals. We correctly classified adults with 83% accuracy and infants with 60% accuracy. We used a multivariate variance components model to estimate how variance in features was partitioned within- and between-individuals. Between-individual variance was the greatest source of variance for all features for adults, and 4/5 features for infants. Despite little sex-specificity and high overlap between duetting adults, the pulse-chirp vocalization is individually distinct. Further, the trills of infants are individually distinct, though to a lesser degree than adults. This study provides evidence for vocal individuality at multiple life stages in a territorial primate.

3:15–3:30 Break

3:30

1pAB8. Long-term foraging dive characteristics of Cross Seamount beaked whales at the Pacific Missile Range Facility. Roanne Manzano-Roth (Naval Information Warfare Ctr. Pacific, 53560 Hull St., San Diego, CA 92152, rmanzano@spawar.navy.mil), Elizabeth Henderson (Naval Information Warfare Ctr. Pacific, San Diego, CA), Gabriela Alongi, Stephen W. Martin, and Brian Matsuyama (National Marine Mammal Foundation, San Diego, CA)

Beaked whale foraging dive clicks similar to those detected at Cross Seamount were detected during passive acoustic monitoring of marine mammals at the Pacific Missile Range Facility (PMRF) off Kauai, Hawaii using bottom-mounted hydrophones from January 2007 through August 2018. The Cross Seamount beaked whale (BWC) foraging dive click frequency sweeps from approximately 18 kHz to above the recording bandwidth of the PMRF sensors of 48 kHz at inter-click-intervals of 0.14 s (std 0.06 s). These unique clicks have only been detected at Cross Seamount and around the Hawaiian Islands, and because they have only been detected at night there has been no visual detections or species confirmation. However, they are distinctive enough to automatically detect, and can be used to describe basic presence at PMRF. Detections averaged 0.047 dives/h (in a 24-h cycle), and occurred almost entirely on the southern part of the range (water depth <1000 m). 702 BWC foraging dives were used to determine a foraging dive baseline over a 12-year period, to compare foraging dive rates during US Naval exercise events and baseline periods, and to contrast BWC foraging dive behavior from Blainville's beaked whale foraging dive behavior at PMRF.

3:45

1pAB9. Passive acoustic beaked whale surveys using gliders off Southern California. David K. Mellinger (Coop. Inst. for Marine Resources Studies, Oregon State Univ., 2030 SE Marine Sci. Dr., Newport, OR 97365, David.Mellinger@oregonstate.edu), Sharon L. Nieukirk, and Selene Fregosi (Coop. Inst. for Marine Resources Studies, Oregon State Univ., Newport, OR)

A passive acoustic survey off Southern California is being conducted in summer–fall 2019 to estimate the distribution of beaked whale species. Two gliders are being flown to estimate the occurrence of Cuvier's, Blainville's, Stejneger's, Hubbs', and Baird's beaked whales in several regions: (1) near the Navy's Southern California Offshore Range (SCORE), in particular near fixed passive recorders there; (2) over the farther reaches of the continental shelf and shelf slope; and (3) over deep (>3000 m) waters offshore of the shelf slope up to 350 km from shore. The gliders collect passive acoustic data up to ca. 90 kHz, which should enable differentiation of these species of beaked whales in situations of reasonably high signal-to-noise ratios. Preliminary results of the glider flights and data analysis will be presented. [Funding from Navy NAVFAC.]

4:00

1pAB10. Inter and intra specific variation in echolocation signals among odontocete species in the Northwest Atlantic, the Temperate Pacific and Hawaii. Tina M. Yack (EcoSound BioAcoust., 9423 Saint Andrews Dr., Santee, CA 92071, tina.yack@ecosoundbioacoustics.com), Julie N. Oswald (Univ. of St. Andrews, Encinitas, CA), Kerry Dunleavy (Bio-Waves, Inc., Encinitas, CA), and Danielle Cholewiak (Passive Acoust. Res. Group/Protected Species Branch, NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA)

Odontocete species use echolocation signals (clicks) to forage and navigate. The aim of this study is to explore inter- and intra-specific variation in clicks among odontocete species in the Northwest Atlantic, Temperate Pacific, and Hawaii. Clicks were examined for seven species of odontocetes—bottlenose dolphins, common dolphins, striped dolphins, rough-toothed dolphins, pilot whales, Risso's dolphins, and Cuvier's beaked whales. Specially developed PAMGuard tools were used to automatically measure a suite of click parameters. Seven parameters were compared within and between species; duration, 10 dB bandwidth, 3 dB bandwidth, center frequency, peak frequency, sweep rate, and number of zero crossings. Significant differences in duration, center, and peak frequency were evident between species within study areas (Dunn's test with Benjamini-Hochberg adjustment <0.05). Geographic variation in click parameters between the three study regions was also examined. Results showed statistically significant pair-wise differences between geographic regions for almost all echolocation click parameters and species (Dunn's test with Benjamini-Hochberg adjustment <0.05). Results suggest species-specific differences in clicks among odontocetes and indicate that geographic variation exists for multiple species. The ecological significance of these findings will be discussed along with implications for classifier development.

4:15

1pAB11. Applications for marine mammal studies using passive acoustic data from the Coastal Endurance array off Newport, Oregon. Elizabeth Ferguson (Ocean Sci. Analytics, 13328 Sparren Ave., San Diego, CA 92129, eferguson@oceanscienceanalytics.com)

The National Science Foundation-funded Ocean Observatories Initiative (OOI) maintains a series of coastal and oceanic monitoring sites that consists of a multitude of physical and biological sensors. As part of this program, OOI has collected continuous, broadband passive acoustic data along the continental shelf and slope off Newport, Oregon since 2016. The Coastal Endurance cabled array consists of two mooring lines that straddle the Columbia River plume, and capture data from this nutrient rich upwelling site along the northeast Pacific coast. In addition to this instrumentation, underwater gliders conduct regular transects to provide better spatial resolution of coastal ocean parameters in the region. Representative data from the continental slope and seamount recording sites of the Coastal Endurance Array were reviewed for vocally active marine mammals. Physical oceanographic variables collected concurrently from the mooring line instrumentation and coastal gliders were used to spatially and temporally characterize the regional marine mammal habitat. This effort demonstrates applications for use of OOI's accessible, multi-instrument platform for conducting marine mammal ecosystem studies.

4:30

1pAB12. Locating calling mammals with signal times amongst shadows and black holes in two-dimensional models. John Spiesberger (Earth and Environ. Sci., Univ. of Pennsylvania, 240 S. 33rd St., Philadelphia, PA 19104-6316, johnsr@sas.upenn.edu)

Calling mammals and other objects were commonly located during the last century with two-dimensional (2-D) models from measurements of a signal's Time Differences of Arrival (TDOA) when the objects are not on the 2-D surface. The common method for locating signals with 2-D models takes signal speed as constant and location is derived by intersecting hyperbolas. However, when correct locations are required, the speed used to derive location must depend on geodesic distance along the 2-D model surface between the object and instrument. For example, when this distance is zero, the speed needed for correct location must also be zero. The dimension reduction from three to two introduces large errors in 2-D models both near and far from the instruments unless variable speeds induced by the dimensional reduction are accounted for. Consequently, methods are derived for generating extremely reliable confidence intervals for locations in 2-D models and identifying regions of the 2-D model where a 3-D model is needed. Because speeds needed for correct location are spatially inhomogeneous in the extreme, isodiachrons emerge as a natural geometry for interpreting location. These issues are caused by choice of coordinates. Similar phenomena occur when coordinate transformations reapplied in relativity [Preprint arxiv:1811.05539 (2018)].

4:45

1pAB13. Investigation of acoustic navigation strategy of bats based on spatial learning by a mathematical model. Yurina Mibe (Life and Med. Sci., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe, Kyoto 610-0321, Japan, ctuc1017@mail4.doshisha.ac.jp), Yasufumi Yamda (Programs of Mathematical and Life Sci., Hiroshima Univ., Hiroshima, Japan), Kentaro Ito (Dept. of Frontier BioSci., Hosei Univ., Tokyo, Japan), Kohta I. Kobayasi, and Shizuko Hiryu (Life and Med. Sci., Doshisha Univ., Kyotanabe, Japan)

Echolocating bats achieve minimal design ultrasonic sensing using one transmitter and two receivers. We have previously found that the bats adaptively change avoidance routes and number and direction of broadcasts as they learn obstacle space by repeatedly flight. In this research, we propose a mathematical model that represents changes in flight path and pulse emission during spatial learning of bats based on these behavioral data. Specifically, by using the psychological effect of alertness to obstacles as a parameter, we confirmed that a simple mathematical model can express the changes in acoustic navigation as the bats learn the space. Furthermore, we flied the bats in spaces where either acoustically clear (chain trains) or poorly permeability obstacles (acrylic boards) were placed in the same layout. As a result, differences in flight speed and pulse direction were found between those two conditions. By comparing the simulation result from mathematical model with behavioral results, we discuss psychological effects on unknown and known space due to differences in acoustic permeability and utilization of spatial memory during navigation. [Work supported by JSPS KAKENHI Grant Nos. JP18H03786 and JP16H06542 and JST PESTO Grant No. JPMJPR14D8.]

5:00

1pAB14. Annular and spiral bubble nets: A simulation-focused analysis of humpback whale feeding strategies. Spencer H. Bryngelson (California Inst. of Technol., 1200 E California Ave., MC104-44, Pasadena, CA 91125, spencer@caltech.edu) and Tim Colonius (California Inst. of Technol., Pasadena, CA)

Humpback whales can generate bubbly regions (called bubble nets) via their blowholes, which they appear to exploit via loud vocalizations for feeding purposes. We model this phenomenon as the acoustic excitation of an dilute bubble net of radially varying void fraction. A fully coupled phase-averaging approach is used to compute the bubble response and corresponding acoustics. We first assess the possibility of a sophisticated wave-guidance behavior of high-frequency whale vocalizations within the bubble net. For a small range of flow parameters, the reflections associated with the bubbly region result in an observable wave-guidance behavior, though even then these reflections disperse rapidly. In light of this, we also consider multiple whales surrounding the bubble net, each vocalizing towards its center. We show that for physically realistic configurations, including variations in the bubble net void fraction and number of whales, the bubble net can keep its core region substantially quieter than the exterior. Finally, we investigate the ability of spiral, rather than annular, geometries for keeping the bubble-free region quiet.

MONDAY AFTERNOON, 2 DECEMBER 2019 — EMPRESS, 1:15 P.M. TO 4:45 P.M.

Session 1pAO

Acoustical Oceanography and ASA Committee on Standards: Observational Acoustical Oceanography: A Look at Enabling Technology from Academia and Industry II

Andrey K. Morozov, Cochair
Marine, Teledyne, 49 Edgerton Drive, North Falmouth, Massachusetts 02556

Orest Diachok, Cochair
Johns Hopkins University APL, 11100 Johns Hopkins Rd., Laurel, Maryland 20723

Chair's Introduction—1:15

Invited Papers

1:20

1pAO1. Long-range target detection and classification system for environmental monitoring at marine hydrokinetic (MHK) sites. Timothy W. Acker (BioSonics, Inc., 4027 Leary Way NW, Seattle, WA 98107, tacker@biosonicsinc.com)

BioSonics and team members are developing a practical, unobtrusive, robust, and cost-effective long range (200–300 m) active acoustic monitoring system to automatically assess marine life behavior at potential or operational MHK sites. The proposed system includes a Perimeter Detector, that will automatically detect and geolocate targets at ranges of 200–300 m. The system will include a Directed Classifier that will be automatically aimed at a detected target to track the target's position in three dimensions. This tracking capability allows automated measurement of the target's behavior (i.e., speed, direction, and tortuosity), a strong indicator of target classification. Acoustic signatures from tracked targets will be analyzed to provide additional target classification information. Low band width, real-time reports will be automatically generated and transmitted to project operators, including target location, depth, behavior, and classification.

1:45

1pAO2. Broadband measurements of the acoustic target strength of mesopelagic fishes. Christopher Bassett (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, cbassett@uw.edu), Andone C. Lavery, and Timothy K. Stanton (Appl. Ocean Phys. and Eng. Dept., Woods Hole Oceanographic Inst., Woods Hole, MA)

The mesopelagic zone (~200–1000 m) is defined by depths where small amounts of sunlight still penetrate but light levels are insufficient to support photosynthetic activity. This region is one of Earth's largest ecosystems and is home to a diverse community of marine animals. Biomass estimates for mesopelagic regions, based on acoustic measurements, suggest small fishes at these depths may dominate total fish biomass. These estimates, however, are generally made using surface echosounder measurements at 38 kHz. These measurements are subject to numerous confounding factors including unknown species compositions, physiology, target strength distributions as a function of depth, calibration, and low signal-to-noise measurements. To address these challenges, in addition to questions that require non-acoustic technologies, a towed, integrated sensor platform, the *Deep-See*, was developed to measure the mesopelagic zone at depth (Lavery *et al.*, 2019). This talk focuses on the acoustics package, which includes commercial and custom split-beam channels from 1–500 kHz. System calibrations and target strength spectra are presented for towed depths up to 700 m, where individual targets are still present in relatively high abundance. Target strength measurements as a function of frequency are considered with an emphasis on their relationship to narrowband, surface-based measurements.

2:10

1pAO3. Teledyne RD instruments engineering developments to increase acoustic measurement range. Paul Wanis (Teledyne RD Instruments, 14020 Stowe Dr., Poway, CA 92064, paul.wanis@teledyne.com)

Acoustic sensors are commonly used for marine measurement and navigation purposes. Acoustic Doppler Current Profilers (ADCPs) measure ocean currents and waves and Doppler Velocity Logs (DVLs) measure instrument motion relative to the sea bed. Both types of instruments perform their measurements by emitting an acoustic signal and measuring the Doppler shift of the return echoes. The maximum sensing range of these instruments is determined by many factors including directivity of the acoustic transducer, spreading losses, frequency-dependent absorption losses, backscatter intensity, and noise in the environment. Users always wish to maximize the range performance in a given application, but there is typically a trade-off between range performance and other parameters such as system size, power, measurement uncertainty, and susceptibility to fading effects. In this presentation, we provide an overview of some of the newest technologies created by Teledyne RD Instruments to improve the achievable range of these instruments. These new technologies include new signal processing algorithms to enhance bandwidth performance while remaining robust against fades and new transducer array technologies to improve the range that can be achieved with a given transducer aperture. These technologies enable greater range performance with a minimal impact to other system parameters.

2:35

1pAO4. Integrative passive acoustic monitoring in Monterey Bay National Marine Sanctuary. John P. Ryan (Res., MBARI, 7700 Sandholdt Rd., Moss Landing, CA 95039, ryjo@mbari.org), Danelle E. Cline, Kelly J. Benoit-Bird, Francisco Chavez, Ben Y. Raanan, Samuel S. Urmy, Yanwu Zhang (Res., MBARI, Moss Landing, CA), William K. Oestreich, Jeremy Goldbogen (Hopkins Marine Station, Stanford Univ., Pacific Grove, CA), John E. Joseph, Tetyana Margolina (Oceanogr., Naval Postgrad. School, Monterey, CA), Andrew DeVogelaere (Monterey Bay National Marine Sanctuary, NOAA, Moss Landing, CA), Karin Forney (Southwest Fisheries Sci. Ctr., NMFS, NOAA, Moss Landing, CA), Raphael M. Kudela (Ocean Sci., Univ. of California, Santa Cruz, CA), and Jarrod A. Santora (Dept. of Appl. Mathematics, Univ. of California, Santa Cruz, CA)

Monterey Bay National Marine Sanctuary, in the highly productive California Current System, is vital habitat for many marine mammal species. Passive acoustic monitoring (PAM) is an effective means to detect species' presence and acoustic behavior. Using the infrastructure of a cabled observatory and detection and classification of sound sources, we examine applications of PAM as an essential part of ecosystem-based research and management. The first case study integrates PAM with multidisciplinary data—whale sighting rates, forage species' abundances, levels of primary production, concentrations of a neurotoxic algal compound, and acoustic modeling results—to examine how occurrence patterns of humpback whale song reflect ecosystem variations. The second case study takes a similar approach to examine blue whale call occurrence, with emphasis on varying detection and classification methods for different call types. The third case study examines an anthropogenic sound source: explosives intended to deter interference of pinnipeds in fishing operations. Toward new developments that are proving insightful, we consider three further approaches: (1) application of unsupervised machine learning methods to advance characterization of humpback whale song structure; (2) integration of PAM and active acoustic sensing to examine predator-prey relationships; and (3) integration of PAM time-series analyses with data from animal tags.

3:00–3:15 Break

3:15

1pAO5. The Lofoten-Vesterålen Ocean Observatory—A cabled infrastructure for operational acoustical oceanography. Geir Pedersen (Technol., NORCE Norwegian Res. Ctr., P.O. Box 6031, Bergen 5892, Norway, geir.pedersen@norceresearch.no), Espen Johnsen (Ecosystem Acoust., Inst. of Marine Res., Bergen, Norway), Lars Alf Ødegaard (Norwegian Defence Res. Establishment, Bergen, Norway), Guosong Zhang, Endre Grimsbø, Gavin Macaulay (Ecosystem Acoust., Inst. of Marine Res., Bergen, Norway), Tor Arne Reinen (Connectivity Technologies and Platforms, SINTEF Digital, Trondheim, Norway), and Anders Hermansen (Equinor, Ranheim, Norway)

In the face of climate change and increasing use of coastal areas, there is a need for high temporal resolution data provided in near real-time, serving research, management, and commercial users. The main objective of Lofoten-Vesterålen (LoVe) Cabled Ocean Observatory is to significantly contribute to the knowledge base of the physical, chemical, and biological environment of the ecologically and economically important LoVe shelf-slope-system. Key sensors for this task are based on acoustical methods for biological and physical oceanography. The majority of the observatory nodes are equipped with state-of-the art scientific echosounders, hydrophones, and acoustic Doppler current profilers (ADCP). Echosounders monitor vertical distribution and density of marine organisms (fish, zooplankton) and biomass flux across the observatory transect. Hydrophones provide continuous monitoring of anthropogenic noise and detection (absence/presence) of vocalizing marine mammals and fish. In addition to the fiber optic communication along the infrastructure subsea cables, each node has the capability of acoustic communication to non-cabled nodes, vessels, and vehicles. The instrument nodes and satellites on which the sensors are mounted feature a range of novel solutions, including technologies contributing to significant reduction in the cost of maintaining the infrastructure compared to traditional cabled infrastructure technologies.

3:30

1pAO6. The case for an Urban Ocean Observatory. Peter J. Stein (Sci. Solutions, Inc., 99 Perimeter Rd., Nashua, NH 03063, pjstein@scisol.com) and Amy Gruhl (Sci. Solutions, Inc., San Diego, CA)

Over the last 16 years the authors have been developing a distributed multi-static active sonar system for port underwater surveillance. Our continuing mission is to develop a robust capability that classifies and tracks the smallest possible objects found in a harbor. We are essentially trying to simultaneously image the entire operational space, and indeed the solution is looking less like a sonar system and more like an imaging system. Developing such a system is a process of continuous optimization, including compensating for the highly time and space variable harbor ocean. Advancement requires greater and greater accuracy and resolution in our environmental measurements and modeling. We envision the system itself becoming a tomographic tool for determining and predicting the ocean state. Here, we make the case for establishing an Urban Ocean Observatory to achieve more rapid and long term progress. Such a laboratory would allow us to efficiently integrate into one test bed, the often stove-piped technologies developed by the underwater acoustic and oceanographic communities. We can study the long term effects of underwater surveillance systems on marine life. The Urban Ocean Observatory can also act as a scale model and speed advancement towards similar littoral and deep water solutions.

3:45

1pAO7. Acoustic determination of CO_2 bubble rise heights during a controlled release experiment. Ann E. Blomberg (Norwegian GeoTech. Inst., Sognsveien 72, Oslo 0855, Norway, aeb@ngi.no), Scott Loranger (Univ. of New Hampshire, Durham, NH), Geir Pedersen (NORCE, Bergen, Norway), Ivar Kristian Waarum, Espen Eek, and Christian Totland (Norwegian GeoTech. Inst., Oslo, Norway)

Carbon capture and storage (CCS), where CO_2 is captured from industrial sources and stored in geological formations, has emerged as a promising method for reducing the amount of CO_2 released into the atmosphere. Current regulations require monitoring the water column above a storage site to verify that there are no indications of leakage. Determining the rise height of CO_2 bubbles is important in order to understand the environmental consequences of a potential CO_2 leak, as well as to establish an efficient monitoring strategy. The rapid dissolution of CO_2 in seawater is likely to limit the rise height of CO_2 bubbles. However, few quantitative studies have been published. During a controlled release experiment in the Oslo Fjord, we simulated a CO_2 leak at a water depth of 60 m. CO_2 bubbles were released at rates of 0.575 l/min and 1.15 l/min, from a 3 mm orifice. Rise heights were determined using multiple echo sounders and sonars mounted on the Simrad Echo R/V. We observed consistent scattering from bubbles as high as 50 m above the leak point. We also experienced that it is challenging to locate the top of the plume using most echo sounders, due to the imaging geometry of the system.

4:00

1pAO8. Acoustic and *in-situ* observations of hydrothermal discharge at ASHES vent field. Guangyu Xu (APL-UW, Seattle, WA), Anatoliy Ivakin (APL-UW, 1013 NE 40th, Seattle, WA 98105, aniv@uw.edu), Karen G. Bemis (Dept. of Marine and Coastal Sci., Rutgers Univ., New Brunswick, NJ), and Darrell Jackson (APL-UW, Seattle, WA)

The Cabled Observatory Vent Imaging Sonar (COVIS) was installed on the Ocean Observatories Initiative's Cabled Array observatory at ASHES hydrothermal vent field on Axial Seamount in July 2018. The acoustic backscatter data recorded by COVIS, in conjunction with *in-situ* temperature measurements, are used to investigate the temporal and spatial variations of hydrothermal venting within COVIS's field-of-view. Areas in which diffuse hydrothermal flow is significant are identified by maps made using the standard deviation of the phase change between pings separated in time by fractions of 1 s. The results demonstrate significant influences of ocean tides and bottom currents on diffuse hydrothermal discharge. Comparison with local seismicity shows a small positive correlation between the areal coverage of diffuse hydrothermal discharge and the seismic activity in the vicinity of the vent field. This finding reveals an intimate connection between hydrothermal activity and geological processes during the dynamic period leading up to the next eruption of Axial Seamount. [Work sponsored by NSF.]

4:15

1pAO9. Application of an embedded general-purpose computing platform to passive acoustic monitoring. Bruce Martin (JASCO Appl. Sci., 32 Troop Ave., Ste. 202, Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com), Art Cole, Craig Hillis, Briand Gaudet, John Moloney, Katie Kowarski (JASCO Appl. Sci., Dartmouth, NS, Canada), David E. Hannay (JASCO Appl. Sci., Victoria, Br. Columbia, Canada), and Emily Maxner (JASCO Appl. Sci., Dartmouth, NS, Canada)

JASCO Applied Sciences has developed a novel passive acoustic monitoring system, the OceanObserver^{TM}, that has been fielded in undersea and surface AUVs as well as on buoys and sub-sea observatories. The hardware is based on the Zynq system-on-a-chip (SoC), which features a large field-programmable gate array (FPGA) and two ARM processing cores. The FPGA is used to implement real-time filtering for the analog-to-digital converters and a real-time detector for odontocete clicks. JASCO's PAMlab JAVA software has been adapted to run in the ARM processors. This allows for the detector algorithms to be tested and proven using recorded data sets, and the deployment of algorithms in the real-time hardware using the same proven software baseline. The advantages of this approach for rapidly adapting generic hardware to diverse monitoring projects will be highlighted.

4:30

1pAO10. Azimuthal, spatial, and temporal variability of acoustic intensity in along-the-shelf direction of Chukchi shelf from June to August 2017. Mohsen Badiey (Univ. of Delaware, Newark, DE 19716, badiey@udel.edu), Lin Wan (Univ. of Delaware, Newark, DE), Sean Pecknold (DRDC - Atlantic, Dartmouth, NS, Canada), and Altan Turgut (Acoust. Div., Naval Res. Lab., Washington, DC)

Spatial, temporal, and azimuthal variability of sound propagation with simultaneously measured oceanography on the Chukchi shelf is reported. Broadband acoustic signals (0.7 to 1.1 kHz) were transmitted from a single

sound source placed near the sound channel axis in 150 m water depth and received by two arrays about 32 km away at different angles forming two acoustic tracks. One was along the 120–150 m isobath and the other crossing 120 to 220 m isobath. The angle between the two acoustic paths was 30 deg. Sound emitted from the common source shows different behavior along each acoustic track. Concurrently detailed water column environmental parameters (i.e., salinity and temperature) were measured in the region in both "along" and "cross-shelf" directions. Emergence of a warm water masses during from late June to July caused a variable channel where transmissions showed markedly large intensity variations (~20 dB). This presentation quantifies data analysis using correlation between acoustics and environmental signals. [Work supported by ONR 322OA.]

MONDAY AFTERNOON, 2 DECEMBER 2019 HANOVER, 1:15 P.M. TO 5:05 P.M.

Session 1pBA

Biomedical Acoustics and Physical Acoustics: Cavitation Nuclei: Bubbles, Droplets, and More II

James J. Kwan, Cochair
School of Chemical and Biomedical Engineering, Nanyang Technological University, 62 Nanyang Drive, Block N1.2, 01-06 Singapore, Singapore

Shashank Sirsi, Cochair
Bioengineering, UT Dallas, 800 West Cambell Rd., Dallas, Texas 75080

Invited Papers

1:15

1pBA1. Biomimetic lung surfactant nanodrops for acoustic droplet vaporization. Alec N. Thomas (Mech. Eng., Univ. of Colorado, Oxford, United Kingdom), Jordan S. Lum, Todd W. Murray (Mech. Eng., Univ. of Colorado, Boulder, CO), and Mark Borden (Mech. Eng., Univ. of Colorado, 1111 Eng. Dr., Campus Box 427, Boulder, CO 80309, mark.borden@colorado.edu)

Acoustic droplet vaporization (ADV) involves the liquid-to-gas phase conversion of a superheated emulsion droplet by ultrasound to form an echogenic bubble. This technology may be useful for medical ultrasound, as nanodroplets small enough to leak through endothelial fenestrations may be converted to echogenic microbubbles for extravascular ultrasound imaging of inflamed and angiogenic vasculature. Additionally, droplets may be transformed to acoustically pulsating microbubbles to enhance ultrasound-guided drug delivery. However, surfactant coverage on the droplet often fails to stabilize the expanding interface during ADV, resulting in transitory microbubbles with limited utility. Here, we show that interfacial melting and spreading by lung surfactant during surface dilation can be harnessed to increase the echogenicity and stability of post-ADV microbubbles. Lung surfactant, whose composition in the mammalian lung has been honed over millions of years of evolution, has thus proven to be a superior coating material for ADV droplets, and its biomimicry will be an important step toward clinical translation of ADV in ultrasound imaging and therapy.

1:35

1pBA2. Acoustic droplet vaporization threshold of perfluorocarbon droplets as a function of frequency: Effects of droplet cores and size. Mitra Aliabouzar (George Washington Univ., Laurel, MD), Krishna N. Kumar (George Washington Univ., Washington, DC), and Kausik Sarkar (George Washington Univ., 801 22nd St. NW, Washington, DC 20052, sarkar@gwu.edu)

Phase shift liquid perfluorocarbon (PFC) droplets vaporizable by ultrasound can be used for many therapeutic and diagnostic applications. The ultrasound activation pressure required for the phase change of these droplets into echogenic microbubbles is termed acoustic droplet vaporization (ADV) threshold. We have studied the ADV and IC (inertial cavitation) thresholds in a tubeless setup using an acoustic means varying frequency of the excitation, the droplet core and their average size. The ADV threshold was found to increase with the increasing frequency for the lowest boiling point liquid, perfluoropentane (PFP), for both large and small average size droplets. For higher boiling point liquids, perfluorohexane (PFH) and perfluorooctyle bromide (PFOB), this study did not detect vaporization for small size droplets at the excitation levels (maximum 4 MPa peak negative) studied here. The large PFOB droplets experienced ADV only at the highest excitation frequency 15 MHz. For large PFH droplets, ADV threshold decreases with frequency that could possibly be due to the superharmonic focusing being a significant effect at larger sizes and the higher excitation pressures. ADV thresholds at all the frequencies studied here occurred at lower rarefactional pressures than IC thresholds indicating that phase transition precedes inertial cavitation.

1:55

1pBA3. Cavitation precursors for biomedical applications. Guillaume Lajoinie (Phys. of Fluids Group, MESA+ Inst. for NanoTechnol. and Tech. Medical (TechMed) Ctr., Univ. of Twente, Drienerloolaan 5, Enschede 7522NB, The Netherlands, g.p.r.lajoinie@utwente.nl)

Stabilized microbubbles are unrivaled as contrast agents for ultrasound imaging owing to its key resonance behavior, arising from the compression of the gas core under ultrasound driving. This resonance also makes them prime candidates for new treatment strategies where microbubbles transport and locally deliver the therapeutics, while mechanically permeating tissues to enhance treatment efficacy. Notwithstanding the achievements made in the field, microbubbles suffer from a strong size limitation, which restricts their access to the circulation and prohibits their use in surrounding tissues, shielded by vessel walls. There is therefore great interest in investigating cavitation precursors that can reach deeper into the target diseased tissues. Such nuclei can be made orders of magnitude smaller than microbubbles without compromising their stability and are thereby allowed to circulate more freely and extravasate beyond the blood vessel endothelium. These precursors present either a liquid form or a gaseous form and can be activated either optically, using laser light, or acoustically using high pressure ultrasound waves. All these features determine the subsequent dynamics of these agents. We recorded the ultrafast cavitation dynamics of a range of agents through high-speed imaging and compare quantitatively to models based on Rayleigh-Plesset-type dynamics.

Contributed Papers

2:15

1pBA4. Standing wave assisted acoustic droplet vaporization for dual payload release from acoustically responsive scaffolds. Mitra Aliabouzar (Univ. of Michigan, 3218-02 Med. Sci. I, 1301 Cathreine St., Ann Arbor, MI 48109, aliabouz@med.umich.edu), Aniket Jivani, Xiaofang Lu, Oliver Kripfgans, Jeffrey B. Fowlkes, and Mario L. Fabiilli (Univ. of Michigan, Ann Arbor, MI)

Ultrasound standing waves have been utilized for many biomedical applications. Here, we demonstrate how standing waves can enhance drug release using acoustic droplet vaporization (ADV), which is the phase-transitioning of perfluorocarbon (PFC) emulsions via ultrasound. These experiments utilized acoustically-responsive scaffolds (ARSs), which are composite fibrin hydrogels containing payload-carrying, monodispersed PFC emulsions. Single- and bi-layer ARSs were generated with dextran-loaded emulsions (diameter: 6 mm) containing either perfluorohexane, or perfluorooctane in each layer. First, we studied the influence of standing waves on payload release from single-layer ARSs. At 4 MPa peak rarefactional pressure, elevated amplitudes due to constructive superposition in the standing wave field enhanced payload release up to 35% at 2.5 MHz in a seven-day longitudinal study. Second, the effect of standing waves was combined with the frequency-dependent ADV to enhance dual payload release from bi-layer ARSs. We demonstrated the sequential release of two dextran payloads from ARSs, which were, respectively, contained within each emulsion, using temporally staggered ADV at 3 MHz (day 0) and 8.8 MHz (day 4). These results will also be discussed in the context of practical strategies for achieving similar conditions for *in vivo* applications

2:30

1pBA5. Perfluorocarbon nanodroplets versus microbubbles in cavitation-enhanced sonothrombolysis of retracted clots. Jinwook Kim (Biomedical Eng., The Univ. of North Carolina at Chapel Hill, 116 Manning Dr., 9018 Mary Ellen Jones Bldg., CB7575, Chapel Hill, NC 27599, jinwookk@email.unc.edu), Ryan DeRuiter, Philip Durham, James Tsuruta, Leela Goel (Biomedical Eng., The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), Xiaoning Jiang (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), and Paul A. Dayton (Biomedical Eng., The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Microbubble-mediated sonothrombolysis has been effective in the treatment of acute clots (<24 h old), where microbubbles provide cavitation nuclei for cavitation-enhanced clot lysis. However, sonothrombolysis has shown very limited efficacy for the treatment of retreated clots (aged >3 days), as retracted clots have denser fibrin network which impedes the permeation of microbubbles. We hypothesize that the phase-change nanodroplets can permeate into retracted clots and improve sonothrombolysis. Phase-change nanodroplets are hundred-nanometer, liquid-filled contrast agents that convert to microbubbles upon excitation by thermal and acoustic energy. Due to their smaller size and higher stability than microbubbles, nanodroplets can provide longer circulation time and improved extravasation into less porous tissues. Here, we compared the thrombolytic effects of lipid shell-decafluorobutane microbubbles and similarly formulated nanodroplets in a flow model that contains a retracted bovine blood clot. Short burst ultrasound (1 MHz, 5.8 MPa, 0.5% duty cycle) was applied in three different therapy scenarios: ultrasound only, ultrasound with microbubbles, ultrasound with nanodroplets (n = 5). We observed internal clot erosion starting inside the clot samples by nanodroplet-mediated ultrasound, whereas the microbubble-mediated case resulted in only surface erosion. The nanodroplet-mediated treatment exhibited an averaged weight loss rate of 3.8 ± 0.1%/min, which was higher than the microbubble-mediated treatment (2.7 ± 0.15%/min).

2:45

1pBA6. A physical model to investigate the acoustic behaviour of microbubbles and nanodroplets within a bone fracture. Sara Ferri (Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, s.ferri@soton.ac.uk), Anastasia Polydorou, Jonathan May (Univ. of Southampton, Southampton, United Kingdom), Qiang Wu, Eleanor P. Stride (Univ. of Oxford, Oxford, United Kingdom), Nicholas D. Evans, and Dario Carugo (Univ. of Southampton, Southampton, United Kingdom)

Impaired fracture healing is a major financial burden for healthcare services; 5%–10% of bone fractures result in non-unions, and there is no clinically approved systemic therapy. This study characterises acoustically stimulated microbubbles (MBs) and nanodroplets (NDs) as non-invasive ultrasound responsive vehicles for the targeted delivery of osteogenic compounds. A microscope-compatible water-tank incorporating a passive cavitation detector was developed to study the acoustic behaviour of MBs and NDs within physical models of bone fractures (gap: 3.5–5.5 mm, angles: 0 deg and 90 deg). The device was designed using COMSOL Multiphysics (Burlington, MA) and tested *in-vitro*. The bone was simulated using a material with comparable acoustic impedance (Sawbones, WA). Numerical simulations showed that the developed experimental set-up generated a relatively uniform acoustic field at a target plane. It could be operated at either 1 or 2 MHz US frequency, at an acoustic pressure in the range 0–1 MPa. The inclusion of a fracture model caused perturbations to the acoustic field, which were dependent on the architecture of the fracture (i.e., relative to the incident US field). Ongoing studies are investigating how these perturbations affect ND/MB response *in-vitro*. Further studies will investigate the relationship between MB/ND acoustic response and the release of biologically active compounds.

3:00–3:15 Break

Invited Papers

3:15

1pBA7. Response of chemically crosslinked microbubble clusters in an ultrasound field. Sugandha Chaudhary, Ronald Hall, Zachary Juan-Sing (BioEng., UT Dallas, Dallas, TX), and Shashank Sirsi (BioEng., UT Dallas, 800 West Cambell Rd., Dallas, TX 75080, Shashank.Sirsi@utdallas.edu)

The purpose of this study is to introduce a new concept of chemically cross-linked microbubble clusters (CCMCs), demonstrate a facile means of their production, and describe how they can potentially be used in biomedical applications. Currently, ultrasound contrast agents (UCAs- also known as "microbubbles") are thought of as single gas bubbles stabilized by a biocompatible shell with sizes ranging from 1–10 μm. By tethering UCAs together into CCMCs, we propose that novel methods of ultrasound-mediated imaging and therapy can be developed through unique inter-bubble interactions in an ultrasound field. One of the major challenges in generating CCMCs is maintaining stability to Ostwald Ripening and coalescence. In this study, we demonstrate that chemically cross-linked microbubble clusters can produce small (<10 μm) quasi-stable complexes that slowly fuse into bubbles with individual gas cores. Furthermore, we demonstrate that this process can be driven with low-intensity ultrasound pulses, enabling a rapid fusion of clusters which could potentially be used to generate novel acoustics. The development of novel microbubble clusters here presents a simple yet robust process for generating novel UCAs with a design that could allow for more versatility in contrast-enhanced ultrasound (CEUS), molecular imaging, and drug delivery applications.

3:35

1pBA8. Possible role of ultrasound research platforms in measuring endogenous decompression bubble nuclei. Virginie Papadopoulou (Joint Dept. of Biomedical Eng., UNC Chapel Hill & NC State Univ., 116 Manning Dr., Chapel Hill, NC 27599, papadopoulou@unc.edu), David Q. Le, and Paul A. Dayton (Joint Dept. of Biomedical Eng., UNC Chapel Hill & NC State Univ., Chapel Hill, NC)

Venous gas emboli (VGE), of the order of several tens of micrometers, can be detected with ultrasound imaging after scuba diving and are a marker of physiological decompression stress. Higher VGE post-dive are associated with a higher risk of decompression sickness, but these bubbles can also be observed after shallow, conservative and completely asymptomatic dives. The risk of decompression sickness, but also the amount of VGE observed post-dive, have been shown to exhibit a significant amount of yet unexplained inter- and intra-subject variability, even for an identical, controlled diving exposure. Importantly, the precise location and formation mechanism of VGE both remain largely unknown. In this talk, we describe the development of ultrasound techniques that overcome barriers to effective imaging of these bubbles in humans. A focus is on optimizing the assessment of post-dive bubble loads by refining their evaluation on echocardiograms to facilitate a more dynamic assessment. Emphasis will then be given to the adaptation of biomedical ultrasound contrast-specific imaging schemes to investigate the infamous "micronuclei" hypothesized to be at the origin of the gas bubbles observed in divers.

3:55

1pBA9. Phospholipid conjugated prodrugs for targeted delivery using ultrasound with microbubbles and nanodroplets. Mendi Marquez, Meghan Hill (Chemical Eng., New Mexico Inst. of Mining and Technol., Socorro, NM), Liliya Frolova (Chemistry, New Mexico Inst. of Mining and Technol., Socorro, NM), and Michaelann Tartis (Chemical Eng., New Mexico Inst. of Mining and Technol., 801 Leroy Pl, Socorro, NM 87801, michaelann.tartis@nmt.edu)

Therapeutic payloads remain a challenge for phospholipid-stabilized microbubbles that are currently used as contrast agents in diagnostic ultrasound imaging. High loading is difficult to achieve due to the metastable phospholipid monolayer that lacks cargo volume, requiring lengthy drug-loaded particle tethering strategies or other sophisticated methods. The purpose of this work is to demonstrate that phospholipid conjugation can anchor chemotherapeutics to the microbubble shell with minimal disruption to phospholipid packing, resulting in an ultrasound theranostic agent that requires minimal preparation prior to administration. Using a Steglich esterification reaction, several potent chemotherapeutics were conjugated to phospholipids and were subsequently incorporated into liposomes, microbubbles, and nanodroplets for biological and particle characterization. Prodrug structures were confirmed with ^{1}H and ^{13}C NMR. Retention of biological activity for each phospholipid prodrug was demonstrated with an MTT cell proliferation assay using liposomes. Loading and stability of nanodroplets and microbubbles were measured with UV-Vis spectroscopy. To demonstrate site-specific delivery potential, these solutions were suspended in a submersible cell culture chamber with adhered HeLa cells. A single element transducer was used to apply a radiation force and fragmentation pulse sequence. Ultrasound-exposed and non-exposed areas treated with prodrug-containing microbubbles and nanodroplets were compared, demonstrating local efficacy.

4:15

1pBA10. Using microbubbles to transduce ultrasound into mechanical deformations capable of activating neurons genetically sensitized to membrane stretch. Stuart Ibsen (Biomedical Eng., Oregon Health and Sci. Univ., 3181 S.W. Sam Jackson Park Rd., Portland, OR 97239-3098, ibsen@ohsu.edu), Ada Tong (The Salk Inst. for Biological Studies, La Jolla, CA), Carolyn Schutt, Sadik Esener (Biomedical Eng., Oregon Health and Sci. Univ., Portland, OR), and Sreekanth Chalasani (The Salk Inst. for Biological Studies, La Jolla, CA)

Ultrasound is an ideal modality to stimulate neurons due to its ability to focus through deep tissue. To facilitate the selective ultrasound activation of neurons within a dense network, we have developed a new method called sonogenetics where we genetically sensitize individual neurons to respond to the mechanical deformations created by an ultrasound pulse. This was done by misexpressing the TRP4 mechanotransduction ion channel in select neurons. As a model system, we used *Caenorhabditis elegans* nematodes which were allowed to freely move on the surface of an agar gel. We found that ultrasound alone did not create enough mechanical deformation at the surface of the agar to activate the TRP4 channels. To overcome this challenge, we introduced stabilized microbubbles to the system by plating them on the gel surface where they naturally surrounded the worms. The interaction between the microbubbles and the ultrasound created mechanical deformations that propagated into the body of the worm and successfully activated the expressed TRP4 causing subsequent neural activation. Activation was confirmed using calcium dependent fluorescent dyes and by quantifying whole worm behavioral changes. This technique can be a valuable tool for future applications in mammalian neural systems aimed at understanding complex neural circuits.

Contributed Papers

4:35

1pBA11. Increased resonant frequency of microslit filtered contrast agents. Jeffrey S. Rowan (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, jrowan@UR.Rochester.edu), James McGrath (Biomedical Eng., Univ. of Rochester, Rochester, NY), and Marvin M. Doyley (Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY)

Imaging using ultrasound contrast agents is highly dependent on the resonant frequency of the population. According to the Marmottant model, modern lipid coated agents have three regimes of resonant frequencies corresponding to the three regimes of bubble dynamics, the buckled, elastic, and ruptured states. The transition from buckled to elastic regime corresponds to the largest change in resonant frequency due to the addition of a second term inversely proportional to the bubbles' radius. For a 1.75 μm bubble, this corresponds to a change from approximately 1.7 MHz, to approximately 5 MHz, leading to higher image resolution, and better separation between the fundamental and subharmonic components. Here we present a novel ultrathin silicon nitride membrane for this purpose. The membrane itself contains thousands of 1.75×50 μm slits and is housed in a centrifuge device. While the exact mechanism is unknown, bubbles that are forced across the membrane during centrifugation show a higher resonant frequency than their strictly size isolated counterparts. In addition, the centrifuged agent also had a lower subharmonic threshold compared to native and size isolated agents, independent of concentration. The possibility of tuning the device to precise frequencies for optimized imaging is also examined.

4:50

1pBA12. The vibration behavior of submicron gas vesicles in response to acoustic excitation as determined via laser doppler vibrometry. An Huang (Material Sci. and Eng., Univ. of California San Diego, 9500 Gilman Dr., SME Rm. 320, La Jolla, CA 92093, anh081@eng.ucsd.edu), Shuai Zhang (Material Sci. and Eng., Univ. of California San Diego, La Jolla, CA), Avinoam Bar-Zion (Div. of Chemistry and Chemical Eng., California Inst. of Technol., Pasadena, CA), Jiaying Wang, Oscar Mena (Dept. of NanoEng., Univ. of California San Diego, La Jolla, CA), Mikhail G. Shapiro (Div. of Chemistry and Chemical Eng., California Inst. of Technol., Pasadena, CA), and James Friend (Dept. of Mech. and Aerosp. Eng., Univ. of California San Diego, La Jolla, CA)

Gas vesicles (GVs) are remarkably stable nano-sized gas-filled protein shells proven effective in ultrasonic imaging. The many potential benefits of GVs arise from their strong gas equilibrium at a submicron size as produced by bacteria or algae, producing significant contrast in ultrasound imaging. The actual vibration behavior of GVs, including buckling and collapse, is poorly understood since the GVs are too small for observation methods of sufficient speed to produce details of the GV deformation during exposure to ultrasound. Traditional optical or acoustic microscopy methods are, in any case, not useful, and *ex-situ*transmission electron microscopy produces useful images but without sufficient time resolution. We propose to instead use laser Doppler vibrometry (LDV) to observe the vibration behavior of GVs. Employing interferometry, LDV offers a far better spatiotemporal resolution. While the typical GV is smaller than 1 μm, an agglomeration of GVs may be used with the LDV to produce a measurable displacement response from a controlled, acoustically delivered pressure. In this talk, we report the fundamental and first harmonic resonance frequencies of GVs and vibration to buckling and collapse at the clinically relevant frequency of 6.5 MHz. We also compare these results with predictions from classic theories of bubble and particle oscillations and finite difference-based computations.

Session 1pMUa

Musical Acoustics: Asian Musical Instruments

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

Chair's Introduction—1:45

Invited Papers

1:50

1pMUa1. The acoustics and culture of the Balinese gamelan. Kent L. Gee (Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu) and Jeremy Grimshaw (Brigham Young Univ., Provo, UT)

The ornateness of Balinese culture is manifest in their percussion ensembles known as gamelan, both in the instruments' design and in their sound. *Ombak* (meaning "wave") well describes the music's shimmering quality that results from acoustic beating. This presentation first summarizes the gamelan, its music, and its role in Balinese culture and then describes collaborative research conducted on a *gamelan semara dana* that led to an improved understanding of the acoustics of its large gongs and metallophones.

2:10

1pMUa2. The journey of the Thai musical instruction collection at the University of California, Los Angeles. Supeena I. Adler (Ethnomusicology, UCLA, UCLA, Los Angeles, CA 90095, supeena@gmail.com)

An ensemble of Thai classical instruments was first brought to UCLA by Dr. David Morton who taught in the Ethnomusicology Department. After he left UCLA, the instruments were disused for decades. In 2015, the author was hired to examine and restore the collection to a playable condition, and subsequently to re-establish the Music of Thailand ensemble class. With financial support from the Thai consulate and the Center for Southeast Asian Studies, the course is now offered regularly and the instrument collection has expanded further. The oldest of the instruments at UCLA originated from the family of a prominent teacher and are highly regarded by their descendants and considered spiritually powerful. These connections were honored with a *wai khruu* ('honoring the teachers') ceremony held at UCLA featuring a professional ensemble of musicians from Thailand. In this paper, the author will introduce the classical instruments of Thailand, their basic construction and tuning, and the process of restoring the collection at UCLA.

2:30

1pMUa3. The khaen: Musical traditions and contemporary innovations. Christopher Adler (Music, Univ. of San Diego, 5544 Forbes Ave., San Diego, CA 92120, adler@alum.mit.edu)

The *khaen* is a bamboo free-reed mouth organ prominent among people of Lao ethnicity in Laos and Northeast Thailand. In Thailand, the *khaen* is considered emblematic of the rural Northeast, but has also enjoyed periods of broader popularity in Thailand, including within the Thai royal courts in the mid-19th century. After a period of modernization during which traditional music was in danger of disappearing, the *khaen* is once again celebrated as a symbol of Northeast identity, taught in public schools, and figures into regional popular music. The polyphonic music of the *khaen* is traditionally played solo and to accompany a solo singer, employing improvisation within five different melodic modes. Recent institutionalization has placed a greater emphasis on composition and playing in ensembles with contemporary folk and electrified popular instruments. The author has contributed to the *khaen* becoming an international concert instrument, by composing, commissioning and recording modern notated works by living composers for the instrument. The paper will survey the changing cultural and musical contexts for the *khaen*, with demonstrations of traditional and contemporary playing styles and techniques.

2:50

1pMUa4. Recent acoustical studies of khaen pipes. Will C. Martin (Eng., Univ. of Iowa, 3131 Seamans Ctr., Iowa City, IA 52242, wcmartin@gmail.com), Kali Nash (Phys., Coe College, Cedar Rapids, IA), and James P. Cottingham (Phys., Coe College, Cedar Rapids, IA)

A khaen pipe is an open bamboo pipe of effective length L, with the free reed located at approximately L/4. The physical length of the pipe is greater than L, with two rectangular tuning slots cut into the pipe defining its effective length and resulting in the presence of cylindrical end sections. Properties of individual khaen pipes and collective effects of two or more pipes sounding together have been studied both experimentally and with finite element modeling software. Special attention has been given to the role of the pipe end sections and the presence of coupling between pipes played simultaneously. In addition to the modes of an open cylindrical pipe expected for the main pipe section, the khaen pipe has modes involving either an end section alone or an end section and the main section

together. The presence of these resulting end sections can significantly affect both sound radiation properties and the timbre of the radiated sound. Music for the khaen almost always involves multiple pipes sounding simultaneously, and the conditions for synchronization of slightly mistuned pipes have been studied.

3:10

1pMUa5. Measurements and impulse pattern formulation (IPF) model of phase transitions in free-reed wind instruments. Rolf Bader (Inst. of Musicology, Univ. of Hamburg, Neue Rabenstr. 13, Hamburg 20354, Germany, R_Bader@t-online.de), Simon Linke, and Robert Mores (Design Media Information, Univ. of Appl. Sci. Hamburg, Hamburg, Germany)

Free-reed wind instruments have a free reed attached to a tube, and are a unique instrument of Laos, Thailand, China, or Japan. Three examples, a 5-pipe hulusheng from Yunnan, China and two Laotian instruments, a 6-pipe Hmong and a 14-pipe Lao khaen are compared in terms of their construction and acoustic properties. With normal playing pressure the pipes only sound when a hole in the pipe is closed. Then, the pitch is determined by the pipe length. When the holes are open the pipes do not sound. Still with low blowing pressure some pipes sound near the frequency of the free reed. A sudden phase transition from reed to pipe frequency happens at a certain blowing pressure. On the contrary, the bordun pipe cannot be muted and produces two pitches with a one-hole fingering. Using the Impulse Pattern Formulation (IPF), which assumes the instrument to work with traveling pressure impulses through the pipes which are reflected, damped and act back on the reed again, the fundamental behaviour of free-reed instruments is modelled.

MONDAY AFTERNOON, 2 DECEMBER 2019 CORONET, 4:30 P.M. TO 5:30 P.M.

Session 1pMUb

Musical Acoustics: Concert: Musical Traditions of Thailand

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

As a relatively young nation state consolidated from multiple kingdoms each with complex histories of cultural exchange and domination, Thailand retains a significant cultural and musical diversity. Most broadly, four cultural regions are well-recognized as having distinct musical traditions today: the central Thai region with an elaborate court tradition of multiple ensembles now considered to be Thailand's "classical" music, the North with a folk tradition emphasizing stringed instruments, the Northeast with a popular rural music featuring the bamboo free-reed mouth organ khaen and log xylophone poong laang, and the Southern peninsula with a strong influence of Malay traditions emphasizing oboe and drums. This performance will feature music and dance from Thailand's four regions with an ensemble of artists led by Supeena Insee Adler, Curator of World Music Instruments and Director of the Music of Thailand Ensemble at UCLA, and Christopher Adler, a composer and ethnomusicologist at the University of San Diego.

Session 1pNS

Noise and Physical Acoustics: Quiet Supersonic Flights 2018 II

Jonathan Rathsam, Cochair
NASA Langley Research Center, MS 463, Hampton, Virginia 23681

Larry J. Cliatt, Cochair
NASA Dryden Flight Research Center, P.O. Box 273, Mail Stop 2228, Edwards, California 93523

Chair's Introduction—1:15

Invited Papers

1:20

1pNS1. Sonic boom prediction and measurement analysis methods for certification of quiet supersonic aircraft. Alexandra Loubeau (Structural Acoust. Branch, NASA Langley Res. Ctr., MS 463, Hampton, VA 23681, a.loubeau@nasa.gov), William Doebler (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA), Robbie Cowart (Gulfstream Aerosp. Corp., Savannah, GA), Sandy R. Liu (Federal Aviation Administration, Washington, DC), Yusuke Naka (Japan Aerosp. Exploration Agency, Tokyo, Japan), Juliet Page (Volpe National Transportation Systems Ctr., Cambridge, MA), Robert S. Downs (Volpe National Transportation Systems Ctr., Cambridge, MA), Peter Coen (NASA Langley Res. Ctr., Hampton, VA), Stephane Lemaire (Dassault Aviation, Saint-Cloud, France), Lucas Wade (Graduate Program in Acoust., The Penn State Univ., University Park, PA), and Victor Sparrow (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

New sonic boom noise certification procedures for quiet supersonic aircraft are needed as part of proposed regulations to potentially amend the ban on civil supersonic overland flight worldwide. In preparation for developing these procedures, analyses are performed on an existing N-wave dataset to exercise various methodologies. Six noise metrics (PL, ASEL, BSEL, DSEL, ESEL, and ISBAP) are calculated for the measurements obtained on the ground and aloft above the turbulent boundary layer. Predictions are performed for these same locations under test day conditions and compared to measurements. In addition, predictions are performed under standard day conditions to facilitate assessment without the effects of real-world winds and atmospheric turbulence. Methods for validating prediction tools or correcting for the atmospheric effects are discussed, and limitations of these methods are identified. Ideas on robust methods for implementation in certification procedures are proposed.

1:40

1pNS2. Evaluation of sonic booms measured in D-SEND#2 flight test. Yusuke Naka (Japan Aerosp. Exploration Agency, Tokyo, Japan, naka.yusuke@jaxa.jp), Masashi Kanamori, Hiroaki Ishikawa, and Yoshikazu Makino (Japan Aerosp. Exploration Agency, Tokyo, Japan)

Japan Aerospace Exploration Agency (JAXA) conducted a supersonic flight test named D-SEND#2 to validate its low-boom technology. In the D-SEND#2 test, sonic booms generated by a scaled, unmanned test vehicle designed by applying JAXA's low-boom concept were measured on and above the ground. Due to a unique flight profile, various types of sonic booms, including N-type and shaped waveforms, generated under different flight conditions were obtained. Although the measurements aloft were intended to avoid or weaken the effects of atmospheric turbulence near the ground on the sonic boom signatures, distortions of waveforms were observed even in the sonic boom data measured above the ground. Noise metrics are calculated for the sonic booms measured at different altitudes, and the effects of the atmospheric turbulence are evaluated. The measurements are also compared with numerical predictions.

2:00

1pNS3. Atmospheric vertical wind effects and their impact on sonic boom. Sriram Rallabhandi (Aeronautics Systems Anal. Branch, NASA Langley, Rm. 190-25, Mailstop 442, NASA Langley Res. Ctr., Hampton, VA 23681, sriram.rallabhandi@nasa.gov)

The one-dimensional augmented Burgers' equation has been generally used for nonlinear lossy sonic boom propagation. This equation models the effects of nonlinearities, loss mechanisms such as absorption and dispersion due to molecular relaxation and thermoviscous dissipation, and geometric spreading through ray tube areas including Blokhintzev scaling as well as atmospheric stratification. Traditionally, atmospheric winds are accounted for by using horizontal wind components varying in magnitude as the sonic boom pressure waveform propagates from near the aircraft toward the ground. Vertical wind components, though present in the real atmosphere, are generally ignored because they are usually much weaker compared to the horizontal components. However, for long propagation distances, such as in the case of secondary booms or speeds at or slightly below Mach cut-off, vertical winds could play a role in changing the location and intensity of sonic booms. This work will update sBOOM, an augmented Burgers' equation solver, to include

the vertical component of atmospheric winds. The sonic boom ground signatures and other relevant data will be compared against those obtained ignoring vertical wind components. Such differences will be discussed and documented for cases which may include shaped low-boom as well as strong shock signatures.

2:20

1pNS4. Design of experiments: X-59 sonic thump carpets in the eastern United States. William Doebler (NASA Langley Res. Ctr., M.S. 463, Hampton, VA 23681, william.j.doebler@nasa.gov), Sara R. Wilson, Alexandra Loubeau (NASA Langley Res. Ctr., Hampton, VA), and Victor Sparrow (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

NASA is conducting a series of computational experiments to quantify atmospheric effects on low noise sonic booms. In the current study, simulated cruise nearfield pressure data from NASA's X-59 Quiet Supersonic Technology aircraft were propagated from the aircraft to the ground at four cardinal headings through five years of realistic atmospheric profiles at 30 locations across the eastern USA. Statistical design of experiments was used to select the locations where primary sonic thump carpet widths (CW) and metric levels at the ground were computed. Atmospheric profiles were taken from the Climate Forecast System Reanalysis database, which contains reanalyzed atmospheric profiles four times daily. Decision tree analyses were performed to determine relative importance of predictors for CW and metric levels. Predictors included were latitude, longitude, date, time of day, season, climate, aircraft heading, and ground elevation. Results of this study indicate the propagation resolution needed to adequately characterize the distributions of CW and ground metric data in terms of the necessary separation distance between propagation locations and the total number of atmospheres through which to propagate. This resolution will be used for a follow-on study of simulated X-59 carpets across the entire US mainland.

2:40

1pNS5. Predicting the statistical occurrence of Mach cut-off sounds using a 3-D ray-tracing model and high-resolution weather data. Zhendong Huang (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, zfh5044@psu.edu) and Victor Sparrow (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

Mach cut-off flight is achieved when a supersonic airplane operates in a narrow speed-altitude envelope just above Mach 1, intending that no sonic boom directly impacts the ground. The present work is assessing how often these ground impacts will occur. An improved 3-D acoustic ray-tracing model has been developed. The High-Resolution Rapid Refresh (HRRR) numerical weather prediction model is used for the atmosphere, which generates hourly analysis data and forecasts gridded at 3 km over the contiguous United States. The propagation of sonic boom noise is simulated for various flight paths, flight levels, safety margins, and atmospheres, to calculate the flight cut-off Mach numbers and corresponding ground speeds. The statistical occurrence of Mach cut-off sounds on the ground due to the atmosphere is predicted. [Work supported by the FAA. The opinions, findings, conclusions and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

3:00–3:15 Break

3:15

1pNS6. Improvements and implementation of code for propagating sonic booms in complex urban environments. Kimberly A. Riegel (Phys., CUNY/Queensborough Community College, 222-05 56th Ave., Bayside, NY 11364, kriegel@qcc.cuny.edu), George Seaton (Eng., SUNY/Westchester Community College, Valhalla, NY), and William Costa (Phys., CUNY/Queensborough Community College, Bayside, NY)

Sonic booms around buildings still require further study for them to be well understood. Previously, a combined ray tracing/radiosity program was created in FORTRAN to simulate the behavior of sonic booms around structures. The program was validated against the 2009 SonicBOBS data and some simple urban geometries. The original program produced good agreement, however, substantial drawbacks to the code were evident. It was not user friendly and creating new geometries was very difficult. The code was ported to Python in order make the user experience better which sacrificed the speed benefits of FORTRAN. Substantial modifications to the code were made to improve the speed and it has been validated with the original Fortran simulations and now runs at a comparable computation time to the original. In addition, the method for creating and importing geometries has been updated so complicated geometries can be incorporated into the simulation more easily by the user. The new program will be discussed in detail. In order to test the new software, several input booms were propagated through an urban environment. The characteristics for the boom before it interacted with the city were then compared to see how the urban environment changes the boom. These updates will allow for the future integration of additional functionality including non-homogeneous atmosphere, turbulence, edge diffraction and other factors that are expected to have a significant impact on the resulting booms.

3:35

1pNS7. Ground effects on sonic boom reflection. Ariane Emmanuelli (Laboratoire de Mécanique des Fluides et d'Acoustique, Ecole Centrale de Lyon, 36 Ave. Guy de Collongue, Ecully 69134, France, aemmanuelli@free.fr), Thomas Lechat, Didier Dragna (Laboratoire de Mécanique des Fluides et d'Acoustique, Ecole Centrale de Lyon, Ecully, France), and Sébastien Ollivier (Laboratoire de Mécanique des Fluides et d'Acoustique, Université Lyon 1, Ecully, France)

Impact of a real terrain on the sonic boom signature at the ground has been little studied in the literature. In the current prediction schemes, a flat and perfectly reflecting ground is usually assumed and the reflected boom is obtained by multiplying the incident boom by a constant factor. In this paper, the effects of a non-flat and absorbing ground are investigated. For this, a numerical study is conducted. The 2-D Euler equations are solved in curvilinear coordinates using high-order finite difference schemes. Signatures of the incident boom typical of a classical N-wave and of a low boom are considered. The variability on the waveform at the ground induced by the terrain irregularities is studied for different characteristic length scales of the terrain.

3:55

1pNS8. Perceived annoyance of Mach-cutoff flight ground signatures compared to common transportation sounds. Jonathan Broyles (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, jmb1134@psu.edu), Michelle C. Vigeant, and Victor W. Sparrow (Acoust., The Penn State Univ., University Park, PA)

Supersonic flight over land was prohibited by the FAA in 1973 to avoid sonic-booms that cause civilian annoyance. A potential operational solution to sonic-booms over land is Mach-cutoff flight, which in ideal atmospheric conditions, refracts the sonic-boom upwards thus creating an evanescent sound-field below. The goal of this study was to compare the perceived degree of annoyance of the evanescent signatures to common transportation sounds. Mach-cutoff flight stimuli were generated using recordings from NASA's Farfield Investigation of No-boom Thresholds dataset and transportation sounds were recorded at local interstates, railroads, and airports. Subjective testing was conducted to investigate annoyance and preference using individual comparisons with an absolute scale and multiple comparisons with a relative scale. An ANOVA and a linear-regression model were calculated to analyze the annoyance and preference of the transportation modes. [This research was funded by the U.S. Federal Aviation Administration Office of Environment and Energy through ASCENT, the FAA Center of Excellence for Alternative Jet Fuels and the Environment, Project 42 through FAA Award No. 13-C-AJFE-PSU under the supervision of Sandy Liu. Any opinions, findings, conclusions or recommendations expressed in this material are those of the authors and do not necessarily reflect the views of the FAA.]

Contributed Papers

4:15

1pNS9. Sensitivity of sonic boom propagation throughout a turbulent atmosphere. Roman Leconte (D'Alembert Inst., Sorbonne Univ., Paris, France), Regis Marchiano (D'Alembert Inst., Sorbonne Univ., Institut Jean le Rond d'Alembert, Sorbonne Université, CNRS, 4 Pl. Jussieu, Paris 75005, France, regis.marchiano@sorbonne-universite.fr), Jean-Camille Chassaing, and François Coulouvrat (D'Alembert Inst., Sorbonne Univ., Paris, France)

Sensitivity of sonic boom propagation throughout a turbulent atmosphere is investigated. Three types of boom of same amplitude but different initial shapes: an ideal N-wave, a measured boom (NASA data for F-18) and a "low" boom (C25D mock-up) with increased rise time are studied. The atmosphere is supposed to be a quiescent and isothermal medium with a superposed synthetic velocity field with homogeneous and isotropic statistical properties satisfying a von Kármán energy spectrum. Using the "random field generation method," the flow velocity turbulent field is governed by three independent parameters: a random matrix, an intensity parameter and a scale parameter (turbulence integral scale). The flow velocity is then used as a base flow for a in-house software called FLHOWARD designed to compute the propagation of acoustic shock waves in heterogeneous media. In order to reduce the number of simulations compared to a Monte-Carlo approach, the study is performed within the generalized chaos polynomial (gPC) framework. Various convergence tests have been performed to define the optimal discretization and gPC order. Stochastic evolution of selected metrics along a 1 km distance are investigated.

4:30

1pNS10. Preliminary analysis of the PCBoom software for calculating secondary sonic booms. Kimberly A. Riegel (Phys., CUNY/Queensborough Community College, 222-05 56th Ave., Bayside, NY 11364, kriegel@qcc.cuny.edu), Victor Sparrow, and Trevor A. Stout (Grad. Prog. in Acoust., Penn State Univ., University Park, PA)

Secondary sonic booms, also known as over-the-top sonic booms, are created during every supersonic aircraft flight, but they only reach the ground some of the time depending on atmospheric conditions. They represent the booms that either initially travel upward from the aircraft and refract to the ground or booms that initially travel downwards, reflect from the ground, and then travel upward and back down again to the ground a second time. Historically secondary booms have not been given as much widespread attention as the primary boom which is loudest, but secondary booms limited Concorde's operations near the coast lines and could be important for planned future supersonic aircraft. The purpose of this paper is to provide an overview of an ongoing study to assess the current capabilities of calculating secondary sonic booms using the PCBoom software. Preliminary results will be presented. [Work supported by the FAA. The opinions, findings, conclusions and recommendations expressed in this material are those of the authors and do not necessarily reflect the views of ASCENT FAA Center of Excellence sponsor organizations.]

Session 1pPA

Physical Acoustics: General Topics in Physical Acoustics II

James Sabatier, Chair
University of Mississippi, 850 Insight Park Ave., Ste. 133, Oxford, Mississippi 38677

Contributed Papers

1:15

1pPA1. Capillary pore pressure effects on acoustic to seismic coupling detection of buried containers. James Sabatier (Univ. of MS, 850 Insight Park Ave., Sue. 133, Oxford, MS 38677, sabatier@olemiss.edu)

The effects of rainfall on one type of buried container has been investigated at a test site near the University of Mississippi. Simulated and natural rainfall experiments were carried out over a three-year time period. The measurements allow for weathering effects on the buried container seismic to acoustic ratio detection contrast. The container was a filled 20 l plastic fuel container buried 30 cm in the natural loess soil. Six containers were buried, two were controls, two were exposed to the natural weather and two were used for simulated rainfall experiments. A loudspeaker was used to excite vibrations in the ground and a co-located vertical component geophone and a microphone measured the out of plane seismic/acoustic ratio in the vicinity of the containers. The rain induced wetting phase causes the container's acoustic-to-seismic coupling contrast to increase and resonant frequency to decrease. The water drying phase restores the original frequency responses of smaller amplitude and higher resonant frequency. The phenomenon is cyclical. The capillary pressure in the pore spaces of soil grains reduces with wetting and increases with drying. These pore pressure changes are controlling the soil stiffness and acoustic response of the buried container.

1:30

1pPA2. Nonlinear acoustic landmine detection experiment: Cylindrical drumlike landmine simulant buried in wetted or unwetted pea gravel. Mikaela M. Furman (Phys. Dept., U.S. Naval Acad., 572 Holloway Rd., Annapolis, MD 21402, furmanmikaela@gmail.com) and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD)

A laboratory experiment for studying acoustic landmine detection involves an 8 in. diameter, 4 in. tall drumlike landmine simulant buried two inches beneath the surface of a pea gravel medium in a cylindrical tank (9 in. O.D., 2 in. thick, 24 in. tall). The cylindrical column of gravel is supported on the sides by the concrete wall, and underneath by the thin circular acrylic top plate of the simulant (which has a thick aluminum bottom plate). Above the surface, a fine spraying mister is controlled to wet the gravel at regular intervals. A porous system near the bottom of the tank drains off excess water. For airborne acoustic excitation two loud speakers are driven by an amplified swept tone (50–850 Hz). A laser Doppler vibrometer connected to a spectrum analyzer measures the particle velocity response across the surface. Nonlinear tuning curve responses were measured on the surface to compare the characteristic frequency shift of the resonant peak with amplitude for nonwetted or wetted gravel. Then, at fixed drive amplitude, the tuning curve response was measured at discrete scan locations across the gravel. Therefore, at any specific resonant frequency, the mode shape response of the coupled simulant-gravel system can be measured.

1:45

1pPA3. Infrasound observations from the 2019 Tornado Season. Brian R. Elbing (Mech. & Aerosp. Eng., Oklahoma State Univ., OSU-MAE, Eng. North 218, Stillwater, OK 74078, elbing@okstate.edu), Christopher Petrin (Mech. & Aerosp. Eng., Oklahoma State Univ., Stillwater, OK), Matthew S. Van Den Broeke, and Erik Green (Earth and Atmospheric Sci., Univ. of Nebraska-Lincoln, Lincoln, NE)

Tornado producing storms have been observed to emit infrasound (sound at frequencies below human hearing) up to 2 h before tornadogenesis. Weak atmospheric attenuation at these frequencies allows for long-range detection. Hence, passive infrasonic monitoring could be a method for long-range studying of tornadogenesis as well as tornado characterization. Identifying the fluid mechanism(s) that produce the infrasound production is critical to enable such capabilities, but currently there are insufficient observations to test potential mechanisms. The 2019 tornado season has been extreme, with over 1000 tornadoes in the United States. This includes numerous severe storms within the range of an infrasound array that was deployed at Oklahoma State University to monitor severe storm infrasound. The current work presents infrasound and, when possible, radar analysis of severe storms during 2019. [Work supported by NSF Grant Nos. 1539070 and NOAA NA18OAR4590307.]

2:00

1pPA4. Field test comparison of various acoustic drone detection methods. Alexander Sedunov, Hady Salloum (Sensor Technologies Appl. Res. (STAR) Ctr., Stevens Inst. of Technol., Hoboken, NJ), Darren Haddad (Air Force Res. Lab., Rome, NY), Alexander Sutin (Sensor Technologies Appl. Res. (STAR) Ctr., Stevens Inst. of Technol., Hoboken, NJ 07030, asutin@stevens.edu), Nikolay Sedunov, and Alexander Yakubovskiy (Sensor Technologies Appl. Res. (STAR) Ctr., Stevens Inst. of Technol., Hoboken, NJ)

Acoustics methods for Unmanned Aerial Systems (UAS) detection have several advantages. They are low cost and passive, do not radiate any RF signals, and can provide UAS classification. The disadvantages of acoustic methods include shorter detection distances and susceptibility to acoustic noise. We present the results of field tests of several acoustic UAS detection methods. The tested systems included the Drone Acoustic Detection System (DADS) that was developed by Stevens Institute of Technology, a 16 microphone cross two-tier acoustic array, directional acoustic microphones (parabolic and shotgun microphones), and the OptiNav ACAM 120 acoustic array. The field tests were conducted to investigate the performance of various acoustic systems for UAS detection and tracking ability of various DJI models. Directional microphones and systems with many microphones did not demonstrate a significant advantage over the lower cost DADS that consists of three nodes, with each node having four microphones. Cross-correlation of signals between various microphone pairs is used for finding the acoustic signal direction of arrival. This system provides acoustic target detection and tracking. Classification software discriminates between a UAS and other possible targets, such as airplanes, helicopters, and ground vehicles. [Work supported by Air Force Research Laboratory under Contract No. FA8750-17-C-0190.]

2:15

1pPA5. Optimization of acoustic drone detection based on tests with linear acoustic array. Darren Haddad (Air Force Res. Lab., AFRL/RIGC, 525 Brooks Rd. Rome, NY, Rome, NY 13441, darren.haddad@us.af.mil), Alexander Sedunov, Hady Salloum, Alexander Sutin, Nikolay Sedunov, and Alexander Yakubovskiy (Sensor Technologies Appl. Res. (STAR) Ctr., Stevens Inst. of Technol., Hoboken, NJ)

In a number of field tests for Unmanned Aerial Systems (UAS) acoustic detection, we used several detection systems including a 16 microphone cross two-tier acoustic array, directional acoustic microphones (parabolic microphones and shotgun microphones), and the OptiNav ACAM 120 acoustic array. The improvement of the detection distance using direction microphones and microphone arrays was less than what was estimated based on the system directivity index. A possible limitation of the directional system performance can be due to wind noise, presence of various noise sources and atmospheric turbulence conditions. The turbulence leads to decorrelation of the acoustic signal received by various elements of a microphone array. The UAS and noise signals recorded by the cross two-tier AFRL acoustic array was used for estimation of external noise and UAS signal correlation depending of the distances between microphones. The array is shaped like a two-tier cross, with each tier spanning 3 m and pseudo-logarithmic spacing between the elements. The coherence measurements between various microphone pairs were used for the estimation of the optimal size of a microphone array for UAS detection and optimal sensor separation for the Stevens Drone Acoustic Detection System (DADS). [Work supported by Air Force Research Laboratory under Contract No. FA8750-17-C-0190.]

2:30

1pPA6. Running with speedster superheroes: What they hear, and practical applications regarding observers travelling at supersonic speeds. Trevor W. Jerome (Penn State, PO Box 30, M.S. 3220B, State College, PA 16804, twjerome@gmail.com)

In 1842, Christian Doppler provided the well-known descriptions of frequency shifts perceived by an observer in motion. These fundamental relationships show that listeners moving relative to a sound source at subsonic speeds perceive a shift in frequency that depends on the direction of relative travel. About one hundred years later, the first speedsters (characters with superhuman speed) emerged in fiction media. Since then, dozens of speedsters have come to life in comic books, television, and cinema. The acoustical content hypothetically perceived by such superheroes at supersonic speeds is investigated. Specifically, the implications of an observer passing through a sound wave—with high and low molecular density regions of a fluid medium—under various atmospheric conditions are explored, assuming turbulence could be sufficiently mitigated. Both transonic and extremely hypersonic (>10 times the speed of sound) regimes are considered. Finally, possible practical applications for supersonic travel are examined.

2:45–3:00 Break

3:00

1pPA7. Human heel-contact and toe-off times measured with an in-air 40 kHz Doppler ultrasound. Sabin Timsina (Univ. of MS, 850 Insight Park Ave., Ste.133, University, MS 38677, stimsina@go.olemiss.edu) and James Sabatier (Univ. of MS, Oxford, MS)

In-air Doppler ultrasound is used to measure human gait parameters as a person walks in a hallway. Single element, 10 mm diameter transmit and receive transducers are used for a walk range from 2–10 m in a continuous wave mode at 40 kHz. The person's foot heel-contact and toe-off times are needed to determine the leg's swing phase and double stance times as well as asymmetries between the left and right legs. At the times, the foot velocity is zero, and the other body segments continue to move with non-zero velocities causing Doppler frequency shifts that mask the smaller foot velocity of interest. We currently use an algorithm to fit the measured foot velocity in the Doppler data to a model in order to estimate these times. This model was developed from simultaneously measured video motion capture and Doppler ultrasonic data for walking persons. Less than desirable results are achieved when the ultrasonic measured times are compared to those from commercially available pressure sensitive gait mats. We will present a comparison of the heel-contact and toe-off times measured using beamed transducer arrays and a foot plate sensor made from a pressure sensitive material.

3:15

1pPA8. Directionality of ground-based exploding balloons. Sarah A. Ostergaard (Brigham Young Univ., Provo, UT, sarahaostergaard@gmail.com), Traci Neilsen, and Julio A. Escobedo (Brigham Young Univ., Provo, UT)

The directionality of an explosion should be accounted for when estimating sound power. Our goal is to estimate the directionality of explosions from measurements on an arc not concentric with the origin of the explosion. To learn how to interpret such data, a test was conducted in a grass-covered field using exploding balloons. The balloons were filled with a stoichiometric mixture of oxy-acetylene and when ignited produced acoustic shock waves. The gas-filled balloons were placed in the ground in preformed "craters." The craters were different shapes to hopefully produce different directionalities. Measurements were taken using both circular microphone arrays centered on each of the four crater locations and a single semi-circle array that was not concentric with any of the craters. The goal is to connect the two measurements by including the effective flow resistivity of the ground and determine how to interpret the directionality from data collected from the semicircle setup. This study was in preparation for a later volcano hazards workshop with buried explosives.

3:30

1pPA9. Ground-based exploding balloon noise propagating over a grass-covered field. Margaret G. Smith (Phys., Brigham Young Univ., N283 ESC, Provo, UT 84602, margaretgrace9816@gmail.com), Tracianne B. Neilsen, Julio A. Escobedo, and Sarah A. Ostergaard (Phys., Brigham Young Univ., Provo, UT)

With ground-based, impulsive acoustic sources, the initial blast wave can be followed by additional noise that is difficult to interpret. As outlined by Embelton [*J. Acoust. Soc. Am.* **100** (1996)], a porous ground can lead to creeping, ground, and surface waves. In a recent test on a grass-covered field, the blast noise from exploding balloons had evidence of a secondary arrival that in some cases was larger than the blast wavefront. The balloons were filled with oxy-acetylene gas and placed on the ground or in holes. The balloons were ignited, and the sound from the resulting explosions were measured at distances of 100, 130, and 160 m. At each of these stations, microphones were placed at four heights: 0.01, 1.2, 2.4, and 3.6 m. For every explosion, the blast wave amplitude increases with height, while the secondary arrival amplitude decreases with height. This variation in height can help identify the type of wave responsible for the secondary arrival. For example,the amplitude of surface waves propagating over grass exponentially decay with height. This study will help distinguish different types of acoustic signals produced by ground-based explosions, such as the Volcano Hazards Workshop in 2018.

3:45

1pPA10. Leaf ruptures: Acoustic impulse events in wildland fires. Kara M. Yedinak (Bldg. and Fire Sci., US Forest Service, Forest Products Lab., 1 Gifford Pinchot Dr., Madison, WI 53726, kara.yedinak@usda.gov), Deborah G. Nemens (School of Environ. and Forest Sci., Univ. of Washington, Seattle, WA), Michael J. Anderson (Mech. Eng., Univ. of Idaho, Moscow, ID), and Raquel Partelli Peltrin (Forest, Rangeland, and Fire Sci., Univ. of Idaho, Moscow, ID)

Quantifying wildland fires is of interest to both the fires science and land management communities. Remote sensing of these events has exclusively focused on electromagnetic spectra emissions. However, wildland fires also produce sound. Unraveling their acoustic profile will likely reveal new information unrealized through traditional remote sensing techniques. We start with the "crackling" sounds often associated with burning live vegetation. The data in these acoustic impulse events are rich, yielding information about the specific plants involved. In work presented here, acoustic impulse events are used to tease apart the influences of species, age, and plant

moisture during combustion of live conifer needles. Needles were collected and burned onsite for six species within the Priest River Experimental Forest. Replicate measurements were carried out in order to reduce the influence of individual branches or trees. Moisture of the needles was ascertained just prior to the experiment through both predawn leaf water potential as well as gravimetric fuel moisture. The burning material was recorded at 50 kHz with a $^1/_2$ in. measurement microphone. The acoustic impulse events were isolated and analyzed to determine likelihood of unique character traits. We present results of investigating the potential differences in acoustic signature based on species and age.

MONDAY AFTERNOON, 2 DECEMBER 2019 SPRECKLES, 1:00 P.M. TO 5:25 P.M.

Session 1pSA

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

Christina J. Naify, Cochair
Naval Research Lab., Pasadena, CA 91109

Bogdan Ioan Popa, Cochair
Mechanical Engineering, University of Michigan, 2350 Hayward St., Ann Arbor, Michigan 48109

Invited Papers

1:00

1pSA1. Recent progress in study of dynamically responsive materials with tailorable microstructural geometric nonlinearities. Nicholas Boechler (Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, nboechler@ucsd.edu)

It is known that strongly nonlinear materials exhibit rich acoustic behavior. Studies of such materials have hitherto been conducted on disparate material systems that leverage either intrinsic material nonlinearities (which could also be thought as geometric in origin depending on the scales considered) or microstructural geometric nonlinearities. While the latter group has been shown to be tunable to a limited extent, the capacity to freely transition between types of nonlinearity has remained out of grasp. In this talk, I will provide an overview of our group's recent work to use topology optimization algorithms to construct periodic microstructure geometries that have specified nonlinear constitutive responses, along with our exploration of elastic wave propagation within such materials.

1:20

1pSA2. Architected micro-lattices for wide-band vibration attenuation. Nikhil J. Gerard, Mourad Oudich (North Carolina State Univ., Raleigh, NC), and Yun Jing (North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695, yjing2@ncsu.edu)

Over the past decade, metamaterials have straddled traditional subject boundaries and emerged as futuristic materials with unconventional functionalities. Architecting material microstructure has been shown to facilitate a myriad of applications for sound and vibration manipulation. The progress of this field has thus been mapped by 3-D printing techniques and the precision that such manufacturing techniques can offer. Recently, large area projection micro-stereolithography, a novel additive manufacturing technique has been shown to be capable of fabricating samples with a high structural complexity and smallest feature sizes ranging from a few microns to over tens of centimeters. From the perspective of mechanical wave propagation, this implies a precise control over micro-structure that can be engineered for a variety of applications. In this work, we develop a unique class of elastic metamaterials that are architected for low frequency, wide-band vibration attenuation. The material is made up of three-dimensional micro-lattices that are shown to possess local resonance band gaps which can be precisely tuned via unit cell geometry and the intrinsic material employed for its fabrication. The working of this material is experimentally verified and the associated functionalities that it can facilitate are discussed.

Contributed Papers

1:40

1pSA3. Immersive boundary conditions for meta-material experimentation. Dirk-Jan van Manen (Geophys., ETH Zurich, Sonneggstrasse 5, Zürich 8092, Switzerland, dirkjan.vanmanen@erdw.ethz.ch), Miguel Moleron, Henrik R. Thomsen, Nele Börsing, Theodor S. Becker (Geophys., ETH Zurich, Zürich, Switzerland), Michael R. Haberman (Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX), and Johan O. Robertsson (Geophys., ETH Zurich, Zürich, Switzerland)

In immersive experimentation, a physical experimentation domain is immersed in a numerical simulation such that waves propagate seamlessly from the physical domain into the numerical simulation and vice-versa. The interaction, governed by a novel immersive boundary condition (IBC), takes place in real-time using hundreds of sources and sensors surrounding the medium connected through a low-latency acquisition, compute and control system. IBCs are currently being realized for acoustic and elastic waves. The ability to impose arbitrary IBCs on an experimentation domain presents unique opportunities for research into novel phononic and parity-time symmetric (PTS) meta-materials. We first show how IBCs can be used to realize phononic materials with arbitrary inclusions by imposing 1-D, 2-D, or 3-D periodic conditions on the boundaries of the experimentation domain. Only one or few unit cells have to be constructed to physically create a complete phononic crystal and reproduce its properties if assumed a component of an infinite periodic lattice. Second, we show how IBCs can be used to implement the gain component of a PTS medium. Since the gain is implemented in the numerical simulation it can be arbitrarily adjusted to exactly balance the experimentally realized loss. Numerical validations and initial experimental results will be presented.

1:55

1pSA4. Acoustic wave confinement by chiral waveguide made of Helmholtz resonators. Yun Zhou (Mech. and Aerosp. Eng., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, yuz421@eng.ucsd.edu), Prabhakar R. Bandaru (Mech. and Aerosp. Eng., Univ. of California, San Diego, La Jolla, CA), and Daniel F. Sievenpiper (Elec. and Comput. Eng., Univ. of California, San Diego, La Jolla, CA)

Confining sound and controlling acoustic wave propagation is of significant importance for efficient energy harvesting. We propose an acoustic waveguide that exists at the interface between two metamaterials consist of unit cells based on Helmholtz resonators arranged in opposite chirality, which gives good confinement of sound wave in the air for frequencies within the bulk band gap of the metamaterial. The amplitude of pressure attenuates by 3 dB within 1/3 of the lattice constant. This waveguide is shown to be robust to frequency and spatial disorders in the system, as long as the edge mode is situated in the bandgap. An acoustic circuit was formed by introducing disorders at sharp corners. Our simulations demonstrate that the acoustic impedance at the interface, as defined by the ratio of the local pressure to the sound velocity, is 3 orders of magnitude smaller than the bulk impedance of the metamaterial for the frequency of interest, giving rise to confinement of the sound wave.

2:10

1pSA5. Extended-reacting liners in time-domain simulations for broadband attenuation with flow. Antoni I. Alomar (LMFA, Ecole Centrale de Lyon, 36 Ave. Guy de Collongue, Lyon, Auvergne-Rhone-Alpes 69134, France, antoni.alomar@ec-lyon.fr), Didier Dragna (LMFA, Ecole Centrale de Lyon, Ecully, France), and Marie-Annick Galland (LMFA, Ecole Centrale de Lyon, Lyon, Auvergne-Rhone-Alps, France)

The modeling and optimization of acoustic liners under grazing flow is an on-going research topic with applications in the aerospace, automotive and railway industries. Recently, the push for next generation airliners and low-noise ventilation systems further increase the need for innovative configurations with improved broadband performance, and with limited size. Porous materials and metamaterials are good candidates to achieve these goals. In this work we propose a time-domain numerical modeling of extended reacting liners using an effective medium approach, which allows to characterize complex media such as porous materials and metamaterials through their effective density and compressibility. The linearized Euler equations are solved in the time domain using efficient high-order finite-difference schemes. The calculation of the convolution integrals is avoided through the auxiliary differential equation method. After a validation of the numerical methodology in a 1-D case, we will present the results for various 2-D configurations, with special attention to the impact of flow. A number of experiments in a Kundt's tube and in a wind tunnel at the Ecole Centrale de Lyon are also under way to test the model predictions.

2:25

1pSA6. Observation of asymmetric scattering in acoustic bianisotropic metagratings. Steven R. Craig (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta 30318, Georgia, scraig32@gatech.edu), Xiaoshi Su, Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ), and Chengzhi Shi (Mech. Eng., Georgia Inst. of Technol., Berkeley, CA)

Bianisotropic metagratings realize asymmetric wave transport by coupling pressure and velocity fields to redirect incident plane waves at an arbitrarily designed angle. The coupled fields simultaneously excite monopole and dipole scattering to redirect the acoustic energy from one grating diffraction mode to another. Given the bianisotropic coupling of the pressure and velocity, we systematically design a metagrating with Bloch wavevectors satisfying the wave-grating interaction in reciprocal space. The grating is designed using a finite element method to vary the dimensional parameters of each unit cell to maximize the scattering efficiency of the Bloch wavevector. We perform 2-D spatial Fourier analysis to verify that the scattering properties of the unit cell match the desired wave-grating interaction in reciprocal space. An experimental realization of the bianisotropic grating demonstrates the experimental results match the desired asymmetric scattering fields.

2:40

1pSA7. Measurement and analysis of sound absorption by a metallic composite foam. Mark J. Cops (Boston Univ., 110 Cummington Mall, Boston, MA 02215, mcops@bu.edu), James G. McDaniel (Boston Univ., Boston, MA), Elizabeth A. Magliula, David J. Bamford (Naval Undersea Warfare Ctr. Div., Newport, RI), and Jay Bliefnick (Acentech, Cambridge, MA)

A composite foam consisting of polyurethane foam embedded in metallic foam is fabricated and evaluated for sound absorbing properties. The normal incidence sound absorption coefficient is measured in an impedance tube for three different types of foams including the composite foam, a rigid framed metallic foam, and an elastic framed polyurethane foam. A lumped element model is used to predict the energy dissipation mechanisms, which include viscous and thermal losses, structural losses, and coupling losses at the metal and polyurethane interface. The best performing composite foam increased sound absorption by a factor of 6 (from 0.1 to 0.6) in the low frequency test range (around 600 Hz) and by a factor of 2 (from 0.2 to 0.4) over the entire test frequency range (250–4500 Hz). The composite foam maintains the high static stiffness of the metal foam and exhibits acoustic compliance characteristic of the polyurethane foam. Composites such as those presented in this work are advantageous for engineering applications where combined high stiffness and energy absorption are required.

2:55–3:10 Break

3:10

1pSA8. Doubling the efficiency of ocean wave power with doubly coiled-up acoustic metamaterial. Joonyoung Lee (Korea Sci. Acad. of KAIST, 167, Eunpyeong Tunnel-ro, Eunpyeong-gu, 1104, Seoul 03440, South Korea, junyoug2001@gmail.com), Mincheol Park (Korea Sci. Acad. of KAIST, Seongnam-Si, South Korea), and Jong-Rim Lee (Korea Sci. Acad. of KAIST, Busan, South Korea)

Ocean wave is a potential renewable energy source with stability and abundance. However, its high initial investment requires high energy harvesting efficiency, limiting its application. In this research, the ocean wave is directly amplified by adopting the structure of the acoustic metamaterial. SHOWPAM—A System of High-efficiency Ocean Wave Power with Acoustic Metamaterial – is designed to achieve this amplification. The mathematical analogy between acoustic wave and water surface wave is found. Also, an effective physical property of the solid obstacle, such as a wall, as a medium of water surface wave is investigated. Coiling-up-space metamaterial structure is simplified to a six-variable system and optimized for maximum water surface wave amplitude using COMSOL Multiphysics. Then, SHOWPAM is constructed based on the simulation result, consisting of a central cavity, energy harvester, and optimized metamaterial on both sides of the cavity. A control model was identical to SHOWPAM but didn't have metamaterial. Both models are tested in the wave generating pool, and power generation of each model was measured. SHOWPAM showed 225% average power compared to the control model, proving the capability of metamaterial structure to enhance ocean wave power.

3:25

1pSA9. Performance mechanisms and stability criteria for highly sub-wavelength, broadband nonreciprocal non-local acoustic metamaterials. Nate Geib, Aritra Sasmal, Zhuzhu Wang, Yuxin Zhai (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Bogdan Ioan Popa (Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109, bipopa@umich.edu), and Karl Grosh (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Active acoustic metamaterials have received considerable attention recently due in large part to their ability to generate nonreciprocal wave transmission. We have shown previously how the spatial separation of acoustic sensors and sources can be leveraged to generate highly sub-wavelength, broadband nonreciprocal behavior in an acoustic waveguide. We showed that by transmitting pressure signals from one location along the waveguide to a loudspeaker at another location, we could achieve a peak isolation of 35 dB with a Q_{10dB} factor of just over 0.25. Here, we discuss the mechanisms by which this nonreciprocity is achieved and the conditions required for the system to be stable. We show how conclusions regarding the performance and stability characteristics of an idealized plane wave acoustic model of our system must be adjusted when accounting for the controlling electronics and loudspeaker dynamics used in the experimental testing.

3:40

1pSA10. A single-detector acoustic camera based on space-coiling anisotropic metamaterial. Tianxi Jiang (Univ. of Sci. and Technol. and China, No. 96 Jinzhai Rd., Hefei 230026, China, jtx9402@mail.ustc.edu.cn) and Qingbo He (Shanghai Jiao Tong Univ., Shanghai, China)

Acoustic imaging is important in diverse applications but is limited by hardware complexity, as an acoustic camera relies on a group of detectors to image sound sources. This study uses spatially encoded structures to solve the problem of single-detector planar acoustic imaging at audio frequencies: a space-coiling metamaterial with high anisotropy is proposed to realize a single-detector acoustic camera. This design outperforms the conventional acoustic camera in terms of dimensions, bandwidth, and cost, and is expected to have real impact on engineering in single-detector acoustic imaging.

3:55

1pSA11. Improving airborne far-field ultrasound acoustic imaging systems using high-K acoustic metamaterial. Amirhossein Yazdkhasti (Mech. Eng., Univ. of Maryland, College park, Hyattsville, MD 20740, amiryazd@umd.edu) and Miao Yu (Mech. Eng., Univ. of Maryland, College Park, MD)

Airborne far-field ultrasound acoustic imaging systems allow scanning a large area that light-based imaging systems fail, such as glass walls and smoky environments. To scan the environment with great details, High-frequency acoustic waves should be used, but high-frequency waves attenuate fast, and it leads to low-range imaging. To address this challenge, a high-k acoustic metamaterial is used. This metamaterial is constructed of a subwavelength sequence of air-material layers and compresses the wave in a way that sound pressure increase in a particular frequency. In this research, a Uniform Linear Array consists of 30 microphones that designed to get a real-time 2-D image of the environment. Using this metamaterial increases the imaging range from 6 m to 15 m. It also improves the capability of the system to recognize small obstacles. By using the metamaterial array, an obstacle with a diameter of 1 cm is visible at a distance of 6 m while the smallest recognizable obstacle for the microphone array at the same distance has the diameter of 6 cm. The effect of the obstacle size and distance on image amplitude has been investigated and compensated. It is concluded that using metamaterials do not only increase the signal to noise ratio but also increase the quality of the image of a complex environment which consists of obstacles with different diameters.

4:10

1pSA12. Stabilization of bianisotropic fluid models for time domain simulation. Samuel P. Wallen (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sam.wallen@utexas.edu), Benjamin M. Goldsberry, 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 bianisotropy [*Phys. Rev. B* **96**, 104303 (2017)]. Recent work has shown that, when wavevectors exceed a critical magnitude, the currently employed model for passive Willis fluids possesses a linear instability stemming from the truncation of higher-order coupling terms in the derivation of the constitutive relationships [*J. Acoust. Soc. Am.* **144**(3), 1832 (2018)]. While this instability can be avoided when using frequency domain methods, it poses significant problems for time domain simulations, as short-wavelength numerical noise may grow exponentially. In the present work, we report on methods of stabilizing the Willis fluid model. Specifically, we consider mathematical manipulations of the wave equation derived from the lowest-order constitutive equations, similar to methods that have been explored in continuum approximations of nonlinear spring-mass chains, as well as higher-order coupling terms. Physical interpretations and implications for time domain finite difference and finite element simulations are discussed. [Work supported by the postdoctoral program at ARL:UT.]

4:25

1pSA13. Linear and nonlinear surface waves in two-dimensional, hexagonal close-packed granular crystals. Samuel P. Wallen (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sam.wallen@utexas.edu) and Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Granular crystals (GCs) are ordered arrays of spherical particles in contact, which have been shown to exhibit rich nonlinear dynamics stemming from Hertzian contact interactions. They are typically modeled as discrete networks of rigid body spheres, which may undergo translational and

rotational motion, with Hertzian interactions modeled as effective nonlinear normal and shear springs. Most of the existing literature on GCs concerns one-dimensional chains of spheres, though a growing body of recent work has explored two-dimensional systems in linear and nonlinear regimes, with multiple packing geometries. In a recent analytical study, Rayleigh-like surface waves were shown to exist in the linear regime for two-dimensional GCs with square packing [*Phys. Rev. E* **93**, 023008 (2016)]. In the present work, we report on surface waves in two-dimensional granular crystals with hexagonal close-packed (HCP) geometry. We demonstrate analytically that, in the linear regime, HCP GCs support Rayleigh-like surface waves analogous to the square-packed case, as well as a leaky surface wave mode with quasi-longitudinal polarization. Extensions of the linear surface modes into the nonlinear regime are demonstrated via numerical examples and compared to results from one-dimensional granular chains. [Work supported by the postdoctoral program at ARL:UT.]

4:40

1pSA14. A feasibility study on achieving mode coalescence in acoustic waveguides. Matthew Kelsten (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, mjk308@scarletmail.rutgers.edu) and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

In a previous talk at ASA it was shown that the double root, commonly referred to as an exceptional point (EP), for modal frequencies in a 2-D or 3-D waveguide can exhibit almost perfect absorption over a relatively broad frequency range. The key to the phenomenon is that the wall impedance is such that modes coalesce at a complex-valued frequency. In this talk, we consider how to feasibly achieve the aforementioned wall impedance with the use of simple resonators. Within the frequency ranges supporting scale separation, it can be shown that a wall aligned with resonators can be modelled as an effective surface with a unique admittance. It is with this concept that we arrive at results showing distinct frequencies at which mode coalescence can occur. Strategies capable of deriving the necessary resonators parameters, and therefore, its dimensions are given. Numerical results are shown for reflection and transmission from a waveguide with the tuned effective surface composed of resonators in addition to the theoretical impedance boundary associated with constant EP behavior. Computer simulations and preliminary experimentation of the phenomena are referenced. Future attempts of utilizing more complicated metasurface designs are discussed for next iterations. [Work supported by NSF.]

4:55

1pSA15. Effective impedance of a locally resonant metasurface. Martin Lott (ISTerre, 1381 Rue de la Piscine, Gières 38610, France, martin.lott@univ-grenoble-alpes.fr) and Philippe Roux (ISTerre, Grenoble, France)

We study here a mesoscopic metasurface made of a randomly distributed set of long vertical metallic rods attached to a thin elastic plate. The A0 Lamb wave mode is strongly affected by the low-quality factor forced resonance of the rods. At such resonance, the rods modify the apparent stiffness of the plate. At the corresponding anti-resonance, the rods clamp the plate in the metamaterial region, which induces eigenfrequency band gaps. Between these two resonant and anti-resonant frequencies, the continuum in the values of the effective rigidity change the reflectivity, and thus the corresponding complex impedance of the metasurface in terms of amplitude and phase. In the present study, experimental data are presented to estimate the effective impedance of such a locally resonant metasurface, in agreement with theory and numerical simulation.

5:10

1pSA16. A cellular sound-absorbing metasurface with subwavelength thickness. Alexandru Crivoi (School of Mech. and Aerosp. Eng., Nanyang Technolog. Univ., Nanyang Ave. 50, Singapore 639798, Singapore, acrivoi@ntu.edu.sg), Danylo Lisevych, and Zheng Fan (School of Mech. and Aerosp. Eng., Nanyang Technolog. Univ., Singapore, Singapore, Singapore)

The sound absorbing metasurface efficient in the audible frequency range is proposed. The metasurface has a cellular structure, each cell having a hexagonal shape with a hollow tunnel inside. The wall of the hexagonal cell is sub-divided into six hollow chambers connected to the central tunnel via the six thinner channels of different diameters. The hollow chambers act as Helmholtz resonators providing six different resonant frequencies for each cell. The negative effective bulk modulus property of the metasurface allows full adsorption at the resonant frequencies. By carefully designing the size of the connecting channels we can manipulate the desired sound adsorption frequency range. The thickness of the metasurface is in the range of 0.03–0.1 wavelengths for the sound frequency range between 300 and 1000 Hz. Six absorption peaks are achieved for each unit cell providing broader range of absorption. The hexagonal shape of the unit cell allows full utilization of the metasurface volume by a standard honeycomb tessellation of the cells. By designing a metasurface containing individual cells with different inner channel diameters, the absorption peaks can be multiplied or overlapped and further broaden the frequency range with the absorption coefficient higher than the desired value.

Session 1pSC

Speech Communication: Self-Perception in Speech Production

Caroline A. Niziolek, Cochair
Communication Sciences and Disorders, University of Wisconsin–Madison, 1500 Highland Drive, Room 485, Madison, Wisconsin 53705

Sarah Bakst, Cochair
Communication Sciences and Disorders, University of Madison-Wisconsin, 1500 Highland Ave., Madison, Wisconsin 53705

Invited Papers

1:15

1pSC1. Adaptation to a physical alteration of the vocal apparatus: The effect of visual self-perception on speech motor plasticity. Guillaume Barbier (Université de Montréal, Université de Montréal, École d'orthophonie et d'audiologie, C.P. 6128, Succursale Centre-ville, Montreal, PQ H3C 3J7, Canada, barbier_guillaume@orange.fr) and Douglas Shiller (Université de Montréal, Montreal, PQ, Canada)

Adapting speech movements to novel or perturbed conditions relies critically on the processing of self-produced sensory information. In the context of a physical alteration of the vocal apparatus (e.g., palatal prosthesis), talkers have been shown to adapt following 10–15 min of practice to produce improved acoustic output. It is possible that additional information, such as ultrasound imaging of the tongue, may help talkers adapt even more effectively to such perturbations. Providing visual feedback of the tongue surface in real-time has shown promise in the treatment of speech-sound disorders. However, it remains unclear whether the addition of such visual information will influence speech plasticity on the timescale examined in experimental studies of speech adaptation. Here, we examine how neurotypical talkers adapt to a palatal prosthesis, relying on auditory and somatosensory feedback alone (n = 15), or with the addition of ultrasound feedback of the tongue, either in the mid-sagittal (n = 15) or coronal plane (n = 15). Differences in adaptation performance between the three feedback conditions were observed, both in the patterns of speech adaptation and the learning after-effects. The results indicate that talkers will rapidly integrate visual articulatory information into their control of oral speech movements in order to guide productions towards improved acoustic outcomes.

1:35

1pSC2. The lexical bias in older adults' compensation to altered auditory feedback. Sarah Colby (Psychol. & Brain Sci., Univ. of Iowa, Seashore Hall, Iowa City, IA 52240, sarah-colby@uiowa.edu), Douglas Shiller (École d'orthophonie et d'audiologie, Université de Montréal, Montreal, PQ, Canada), Meghan Clayards, and Shari Baum (School of Commun. Sci. & Disord., McGill Univ., Montreal, PQ, Canada)

Older adults show a larger lexical bias when categorizing speech sounds (i.e., Ganong effect), such that they are even more biased than younger adults to categorize ambiguous tokens as real words rather than non-words (Mattys and Scharenborg, 2014). Using an altered auditory feedback paradigm, we sought to investigate whether this larger perceptual bias would be reflected in older adults' compensation to perturbations of their own speech. Groups of older (n = 27) and younger adults (n = 35) produced monosyllabic words and non-words containing the vowel /ɛ/. Altered auditory feedback lowered the first formant (F1) of the vowel towards an F1 characteristic of /ɪ/. This real-time frequency manipulation shifted the perceived lexical status of the stimuli (i.e., words were shifted towards non-words, and non-words towards words). Younger adults compensated more to non-words that were shifted towards real words (e.g., kess-kiss) than to real words that were shifted towards non-words (e.g., chest-chist), consistent with previous findings (Bourguignon *et al.*, 2014). However, older adults did not show the same pattern of compensation as younger adults, suggesting that different mechanisms are involved in older adults' lexical bias. A combination of age-related cognitive and sensory changes likely influences the effect of lexical status on sensorimotor adaptation in older adults.

1:55

1pSC3. A causal role of superior temporal gyrus in auditory-motor control of vocal production. 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) and Dongxu Liu (Rehabilitation Medicine, The First Affiliated Hospital of Sun Yat-sen Univ., Guangzhou, China)

Accumulating evidence has shown that the superior temporal gyrus (STG) plays an important role in auditory feedback control of vocal production. The precise role of this region in auditory-vocal integration, however, remains unclear. The present event-related potential study applied a transcranial magnetic stimulation (TMS) protocol, continuous theta-burst stimulation (c-TBS), to depress activity in the left STG as young adults produced sustained vowels while their voice was unexpectedly pitch-shifted downwards by 50 or 200 cents. The behavioral results showed significantly larger vocal compensations for pitch perturbations in the c-TBS condition as compared to the sham condition. At the cortical level, c-TBS over left STG led to significantly smaller cortical P2 responses than the sham condition. These findings provide the first causal evidence for linking the STG to auditory feedback control of vocal production. Enhanced vocal compensations for pitch perturbations as a result of disrupting the STG activity lend support to the idea that the STG may be critically involved in top-down inhibitory mechanisms of speech motor control, through which vocal motor behaviors can be appropriately regulated without being excessively influenced by auditory feedback.

2:10

1pSC4. Neural representations of enhanced speech motor control in trained singers. Sheena Waters (Royal Holloway, Univ. of London, London, United Kingdom), Elise Kanber, Nadine Lavan (Speech Hearing and Phonetic Sci., Univ. College London, London, United Kingdom), Daniel Carey (Royal Holloway, Univ. of London, Dublin, Ireland), Valentina Cartei (Univ. of Sussex, Brighton, United Kingdom), Clare Lally (Royal Holloway, Univ. of London, London, United Kingdom), Marc Miquel (Queen Mary Univ. of London, London, United Kingdom), and Carolyn McGettigan (Speech Hearing and Phonetic Sci., Univ. College London, London, United Kingdom, c.mcgettigan@ucl.ac.uk)

Humans are unrivalled amongst the great apes in our capacity for vocal learning, which forms a key component our facility for articulate speech. Here, we used a speech imitation task to investigate the neural representation of the human larynx, which has a distinct cortical topography and innervation in humans that may underpin our sophisticated vocal capabilities compared with non-human primates. In an MRI study, 25 highly trained singers and 24 non-singing control participants adjusted voice pitch (F0) and vocal tract length (VTL) to mimic altered auditory targets generated from recordings of their own speech. Participants simultaneously underwent real-time anatomical scans of the vocal tract and functional scans of brain activity. Representational similarity analyses of neural activation data identified representation of the two vocal parameters in regions of somatomotor cortex previously associated with laryngeal control. Singers showed more accurate task-relevant larynx height modulation (as measured with vocal tract MRI) when imitating VTL, which was underpinned by stronger representation of VTL within a region of right dorsal somatomotor cortex previously related to singing experience. Our findings offer the first behaviourally validated, representational account of speech imitation within larynx somatomotor cortex, suggesting that the behavioral and neural correlates of singing expertise extend to human speech.

2:25–2:45 Break

Invited Papers

2:45

1pSC5. Individuals with cochlear implants use feedback from their own productions to alter the pitch of their voice. Justin Aronoff (Speech and Hearing Sci., Univ. of Illinois at Urbana-Champaign, 901 S 6th St., Champaign, IL 61820, jaronoff@illinois.edu)

The feedback cochlear implant (CI) users' receive of their own voice is greatly distorted by the CI processor, which may cause these individuals to ignore that feedback. A series of studies were conducted to investigate if this is the case. In the first study, because bilateral CI users often hear rumbly sounds with one CI and squeaky sounds with the other CI, bilateral CI users were asked to produce a sustained vowel (/a/) when using either their left or right CI. Many participants produced different F0s depending on which CI they were using. In the second study, participants were presented with altered feedback, where brief shifts in the spectrum were introduced to the feedback from their own voice. Participants shifted the pitch of their voice in response. In the third experiment, stimulation patterns were shifted either apically or basally while participants completed a number of vocal tasks. Participants reliably shifted the pitch of their voice, compensating for the shifted place of stimulation. Taken together, the results indicate that CI users are monitoring their own vocal productions and using that feedback to alter their F0. Additionally, the results suggest that the place of stimulation is a key aspect of that feedback.

3:05

1pSC6. Vowel production in children with myotonic dystrophy: A lip-tube perturbation study. Lucie Menard (Linguist, Universite du PQ a Montreal, CP 8888, succ. Centre-Ville, Montreal, PQ H3C 3P8, Canada, menard.lucie@uqam.ca), Pamela Trudeau-Fisette, Cristina Uribe, and Camille Vidou (Linguist, Universite du PQ a Montreal, Montreal, PQ, Canada)

Myotonic dystrophy, a neurodegenerative disease that causes muscle weakness and difficulties in muscle relaxation after contraction, frequently affects orofacial articulatory dynamics leading to decreased speech intelligibility, particularly in children. We aimed to investigate the effects of myotonic dystrophy on sensorimotor relationships in children's speech through a study of compensations for a lip-tube perturbation. We recruited fourteen 6- to 14-year-old French-speaking children diagnosed with myotonic dystrophy and

14 aged-matched typically developing children. They were asked to produce repetitions of the vowel /u/ with and without a 15-mm-diameter tube inserted between the lips. A synchronized ultrasound, Optotrak motion tracking system, and audio recording system was used to track lip and jaw displacement as well as tongue shape and position. Separate analyses were conducted on the first (F1), the second formant (F2), and the fundamental frequency (F0). Results revealed a significant main effect of group and an interaction between group and condition. Perceptual ratings of the produced vowels suggest that children with myotonic dystrophy rely more on auditory feedback than their typically developing peers. Together, results suggest that auditory feedback plays an important role in speech compensation, especially when the production system is impaired.

3:25

1pSC7. Self-monitoring of speech production in individuals who stutter. Ludo Max (Dept Speech & Hearing Sci., Univ. of Washington, Seattle, WA 98105-6246, LudoMax@uw.edu) and Kwang S. Kim (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

Self-monitoring plays a critical role in the acquisition and refining of all skilled movements, including speech production. In current theoretical models of the neural control of movement, self-monitoring depends on the acquisition and updating of internal representations of the motor-to-sensory transformations that take place from the generation of motor commands to final movement outcomes (e.g., articulatory movements and speech sound output). During movement planning and execution, these representations are used to predict sensory consequences ahead of time, and actual and predicted feedback can then be compared. For speech, auditory-motor learning (the process through which neural representations are learned and maintained) and auditory-motor integration in general (including bidirectional interactions during the online control of speech) are believed to play an important role in persistent developmental stuttering. This presentation will give a brief overview of recent insights into potentially critical differences between children and adults who stutter vs. individuals with typical speech in terms of: (a) auditory-motor learning in the presence of experimentally perturbed auditory feedback; (b) feedback-driven corrections of ongoing unperturbed speech movements; and (c) the relationship between such learning-based or instantaneous articulatory adjustments and a central modulation of auditory processing that already starts during movement planning prior to speech onset.

3:45–4:00 Break

Contributed Paper

4:00

1pSC8. The role of auditory feedback in learning speech of second languages. Xing Tian (NYU Shanghai, 1555 Century Ave., Rm. 1259, Shangai 200122, China, xing.tian@nyu.edu) and Xiaoluan Liu (NYU Shanghai, Shanghai, China)

Learning to speak requires coordination between auditory and motor systems. Motor programs need to be fine-tuned based on goals and feedback. The motor-tuning process includes establishing the sensory-to-motor transformation (generate motor codes from articulation) and motor-to-sensory transformation (generate predictions for speech control). In this study, we investigated the relations among the three factors (feedback, goals, and predictions) in speech production in the context of second language learning. Adult Mandarin speakers were asked to learn non-native vowels: /ø/ and /ɵ/—the former being less similar than the latter to Mandarin vowels, in feedback available or feedback masked conditions. We found no improvement in learning when feedback was masked, suggesting that motor-based prediction could not directly compare with goals for adult second language acquisition. Furthermore, feedback helped to learn only when target sounds are distinct from existing sounds in one's native speech (competition between prediction and goals is minimal). The results suggest prediction and goals may share a similar representational format, which could yield a competing relation in speech learning. The feedback can conditionally overcome such interference between prediction and goals. The study further probed the functional relations among key components (prediction, goals, and feedback) of sensorimotor integration in speech production and learning.

Invited Papers

4:15

1pSC9. Changes in L2 production variability associated with visual biofeedback training. Joanne Jingwen Li (New York Univ., 665 Broadway, 9th Fl., New York, NY 10012, jl6132@nyu.edu), Samantha Ayala, Daphna Harel (New York Univ., New York, NY), Douglas Shiller (Université de Montréal, Montreal, PQ, Canada), Caroline A. Niziolek (Univ. of Wisconsin–Madison, Madison, WI), and Tara McAllister (New York Univ., New York, NY)

Previous research suggests that acoustic variability across repeated productions of a phoneme could index the robustness of speech motor plans. Variability at onset is thought to reflect robustness in feedforward control [1,2], while variability at midpoint may reflect the narrowness of sensory targets and/or speakers' capacity for feedback correction [3]. In L2 production, speakers may show elevated variability at onset because of unfamiliar motor plans and at midpoint due to weak sensory targets [2]. This study investigated variability in L2 vowel production before and after a brief training incorporating visual biofeedback. We hypothesized that midpoint variability would decrease after training, reflecting refinement of the auditory target, whereas onset variability may remain unchanged because the limited training might not be sufficient to establish a robust feedforward plan. Sixty native English speakers received 1 h of biofeedback training to produce two Mandarin vowels (/y, u/). After training, both vowels showed decreased midpoint variability, but only /y/ showed reduced onset variability. In addition, for /u/ only, higher pre-training midpoint variability was correlated with greater improvement in accuracy (distance from native-speaker target). These results are discussed in connection with the differing relationships of Mandarin /y/ and /u/ to the English vowel inventory.

4:35

1pSC10. The role of auditory feedback in error-detection and correction in first and second language production. Sarah Bakst (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, 1500 Highland Ave., Madison, WI 53705, sbakst@wisc.edu) and Caroline A. Niziolek (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, Madison, WI)

Speakers self-correct small variations they hear in their own speech, whether self-produced or experimentally induced perturbations to auditory feedback. In a first language (L1), phonetic categories and therefore auditory expectations are well-established, but in a second language (L2), lack of experience with new categories may interfere with the ability to detect and correct such deviations. We present two companion studies to test how English speakers learning French detect such variations in their auditory feedback in both L1 and L2. In one study, speakers uttered monosyllabic English and French words while simultaneously receiving different levels of masking noise to determine how access to auditory feedback information impacts speakers' ability to make online adjustments to self-produced speech. There was greater acoustic variability at the midpoint of utterances in L2 than in L1, regardless of noise condition, consistent with a differential use of auditory feedback information in the two languages. In a second study, speakers received altered auditory feedback. On one-third of trials, the vowel formants of produced words were shifted up or down, creating the perception of an error. We will compare correction of internally and externally generated error to understand speakers' sensitivity to auditory feedback in native and learned languages.

Contributed Paper

4:55

1pSC11. Perception precedes production in native Mandarin speakers of English. Madeleine Yu (Cognit. Sci., UC San Diego, Toronto, ON, Canada), Reina Mizrahi (Cognit. Sci., UC San Diego, Chula Vista, CA), and Sarah Creel (Cognit. Sci., UC San Diego, 9500 Gilman Dr., MC 0515, La Jolla, CA 92093, screel@ucsd.edu)

Nonnative accents are commonplace, but why? Ample research shows that perceptual representations of second-language speakers are shaped by their first language. But is production also affected? If perceptual representations perfectly control motor production, then second-language speakers should understand their own speech accurately. To test this, we recorded 48 native Mandarin speakers labeling pictures in English. We then played back their own recorded productions (e.g., "lock") as they chose one of four pictures (lock, log, shape, and ship). They also heard a paired native English speaker. Words contained contrasts challenging for Mandarin speakers, principally coda voicing (*lock, log*) and similar-vowel (*shape, ship*) pairs. Listeners achieved 89% accuracy on both their own productions and native speakers,' suggesting good matching between perception and production. However, errors were unevenly distributed: Mandarin speakers heard their own voiced codas (*log*) as voiceless (*lock*) more often than the reverse (10% vs. 5%, $p = 0.002$). This mirrors a similar but larger voiceless bias in native-English listeners hearing accented stimuli, suggesting that Mandarin speakers' coda voicing perception is more nativelike than their production. Ongoing work attempts to differentiate interlanguage intelligibility effects from learning of idiosyncratic speech patterns, and we are exploring which acoustic features predict recognition.

Session 1pSP

Signal Processing in Acoustics, Architectural Acoustics, and Noise: Signal Processing for Architectural Acoustics and Noise Control II

Matthew S. Byrne, Cochair
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Kainam Thomas Wong, Cochair
Beihang Univ., School of General Eng., Beijing 100083, China

Invited Papers

1:00

1pSP1. Time reversal focusing of high amplitude sound in a reverberation chamber: Optimizing the placement of transducers. Brian D. Patchett (Phys. & Astronomy, Brigham Young Univ., 800 W University Pkwy, MS-179, Orem, UT 84058, brian.d.patchett@gmail.com) and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Time reversal (TR) is a signal processing technique that may be used to intentionally generate high amplitude focusing of sound. The use of time reversal in room acoustics has been studied previously, but an analysis of the dependence of spatial positioning of the transducers in the room, and its effect on high amplitude focus signals has not previously been explored. Experiments are carried out in two reverberation chambers of differing size with eight compression drivers. Using these drivers, the impulse response is found and reversed in time. The spatial dependence of the transducers position is measured by calculating the time reversed impulse response at various transducer positions and then measuring the peak focal amplitude generated when the time reversed signal is broadcast to those positions. Multiple measurements suggest that the amplitude of the focus signal changes with both room size and proximity to a reflecting surface in the chambers.

1:20

1pSP2. Performance comparison of various strategies in 24-channel versatile sound field reproduction system. Akira Omoto (Faculty of Design, Kyushu Univ., Shiobaru 4-9-1, Minami-ku, Fukuoka 815-8540, Japan, omoto@design.kyushu-u.ac.jp) and Hiroshi Kashiwazaki (Graduate School of Design, Kyushu Univ., Fukuoka, Japan)

A sound field reproducing system which consists of a hedgehog-shaped 24-channel narrow directional microphone and the speakers of the same number is currently proposed. The system is aimed for the versatile usage, for example, the reproduction of the sound field in a concert hall, and also applicable to various simulation for noise control engineering. The basic reproduction technique is intuitive and straightforward, in which a signal recorded by a particular microphone would be emitted from a loudspeaker located at the almost same direction. To compensate for the insufficient directional characteristics in the low-frequency range, some kinds of signal processing are necessary. So far, several methods, such as the inverse filtering based on the boundary surface control principle, the ambisonics, the beamforming, are attempted. The processed low-frequency components are integrated with the directly assigned high-frequency components. In addition to the procedure that uses 24 recorded signals, the alternative method which reproduces the directional information by convolving the measured directional impulse responses with the signal of the fewer channel, such as monaural or stereo format, could be used. The performance of these various reproduction methods is compared with each other.

1:40

1pSP3. The use of an acoustic quiet zone as an active acoustic cloaking system. Charlie House (Inst. of Sound & Vib. Res., Univ. of Southampton, Bldg 13 - ISVR, Highfield Campus, Southampton, Hampshire SO171BJ, United Kingdom, c.house@soton.ac.uk), Jordan Cheer, and Stephen Daley (Inst. of Sound & Vib. Res., Univ. of Southampton, Southampton, Hampshire, United Kingdom)

The use of active control to acoustically cloak an object has been demonstrated previously, and is effective if the scattered component of the pressure field can be measured and directly minimised. In practice, this is non-trivial as a pressure sensor in the sound-field will detect the superposition of the incident and scattered pressures. An alternative approach is presented here which uses the control sources to generate a zone of quiet around the scattering object, with a constraint on their exterior radiation. This reduces the incident acoustic pressure on the object and thus the acoustic scattering. In this case, real-time measurements of scattered pressure or detailed geometrical knowledge of the scattering object are not required. The performance of the proposed acoustic cloaking strategy is assessed using simulated data of a rigid spherical scattering object, using a practical arrangement of control sources and error sensors. The same arrangement of sources and sensors is also used to directly minimise the scattered sound field

2:00

1pSP4. Active control of plane waves: Transmission, reflection, absorption and steering. Jordan Cheer (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., Highfield, Southampton, Hampshire SO17 2LG, United Kingdom, j.cheer@soton.ac.uk)

Active control of an incident acoustic wave using an array of secondary source has been extensively investigated. In general, the objective is to attenuate the propagating wave so as to generate a space with a reduced sound pressure level. However, there are a variety of applications where more flexible manipulation of the acoustic wave propagation is required. For example, in architectural acoustics it may be desirable to achieve a particular level of absorption rather than total cancellation of the acoustic wave. This paper presents an investigation into the use of an array of secondary sources to control the transmission, reflection, absorption and steering of an incident plane wave. A formulation is presented that enables the secondary sources to be driven to achieve the desired levels of transmission, reflection and absorption, as well as control over the angle of wave propagation. The requirements in terms of the secondary sources, error sensors and reference sensors are discussed and the physical limits on the control performance are evaluated.

2:20

1pSP5. Selective active noise cancellation based on directional reference signals. Buye Xu (Facebook Reality Labs, Facebook, 8747 148th Ave. NE, Redmond, WA 98052, buye_xu@oculus.com) and Tony Miller (Facebook Reality Labs, Facebook, Redmond, WA)

Active noise control (ANC) technologies have been successfully applied in headphones to cancel the sound that propagates into one's ear canal from outside. However, the standard ANC systems would attenuate the sound from all the sound sources blindly ignoring that a listener may prefer to listen to some of them. This paper proposes a feed-forward ANC system for ear-level applications, which is capable of preserving the sound of a source in any given direction relative to a listener while attenuating the sound in other directions. The selective cancellation is achieved based on the creation of appropriate reference signals from the output of a microphone array wore on user's head. Theoretical analysis and simulation results will be presented to demonstrate the potential benefit and limitations of such a system.

2:40–3:00 Break

3:00

1pSP6. Effects of room acoustics on talkers: A use case for speech-acoustic signal processing in architectural acoustics. Michael Rollins (Biomedical Eng., Univ. of Cincinnati, MSB 6303, 231 Albert Sabin Way, Cincinnati, OH 45267, rollinmk@mail.uc.edu), Timothy W. Leishman (Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Eric J. Hunter (Communicative Sci. & Disord., Michigan State Univ., East Lansing, MI)

In the past, several room-acoustic parameters, recommendations, and standards have been developed with emphasis on accommodating the listener. Recent research has shown that talkers also respond to room acoustics via vocal adaptations. In fact, room acoustics may influence the development of voice disorders in occupational speech users. To evaluate the relationships between room acoustics and vocal parameters of healthy talkers, the authors of this presentation analyzed speech-acoustic signals under a variety of acoustical conditions. Vocal parameters were derived from recordings, and statistically significant effects of room acoustics were verified using mixed-model analysis of variance tests. Changes in reverberation time (T20), early decay time (EDT), clarity index (C50), speech transmission index (STI), and room gain (GRG) all showed highly correlated effects on certain vocal parameters, including speaking level standard deviation, speaking rate, and acoustic vocal quality index. The results show that as T20, EDT, and GRG increased, and as C50 and STI decreased, vocal parameters tended toward dysphonic phonation. These findings increase our understanding of the impact of room acoustics on vocal production, informing room design to help mitigate unhealthy vocal exertion and, by extension, voice problems. The study provides examples of how signal processing of speech signals benefits architectural-acoustics knowledge.

3:20

1pSP7. A study on attention-based objective function in deep denoising autoencoder based speech enhancement. Yi-Ying Kao, Hsiang-Ping Hsu (Investigation Bureau, Ministry of Justice, R.O.C, Taipei, Taiwan), Kuo-Hsuan Hung (Academia Sinica, 128 Academia Rd., Section 2, Nankang, Taipei 11529, Taiwan, khhung@iis.sinica.edu.tw), Shih-Kuang Lee (Academia Sinica, Taipei, Taiwan), Ying-Hui Lai, Chen-Yu Chiang (National Taipei Univ., Taipei, Taiwan), and Yu Tsao (Academia Sinica, Taipei, Taiwan)

Speech is one of the most direct and convenient human\machine interfaces. In real-world scenarios, however, various interferences and noises may deteriorate the speech signals and thus reduce speech quality and intelligibility. Therefore, speech enhancement (SE) is an essential component in speech-communication systems. Recently, numerous deep-learning-based SE approaches have been proposed and yield satisfactory performance. In a deep-learning-based SE system, defining a proper objective function plays a crucial role to its success. Generally, the mean square error (MSE) of the predicted and desired outputs are used to form the objective function to learn the parameters in deep-learning models. Because a sequence of speech signals contains various patterns, such as consonant, vowel, beginning and ending silences, and short pauses, it is not optimal to simply use MSE as the objective function, since the contributions of these different patterns may be averaged out. Instead, we should apply specific weights for distinct patterns when designing the objective function. In this presentation, we present a novel objective function, which is used in deep denoising autoencoder-based SE system. The proposed objective function is derived by MSE with multiplying a ratio calculated from clean and noisy speech. The result is evaluated using standardized evaluation metrics, and experiment results confirm the proposed objective function is beneficial to improve the intelligibility of enhanced speech.

3:40

1pSP8. The spectral structure of acoustic time series can predict the perceptual assessment of urban soundscapes. Andrew Mitchell (Inst. for Environ. Design and Eng., Univ. College London, Central House, 14 Upper Woburn, London WC1H 0NN, United Kingdom, andrew.mitchell.18@ucl.ac.uk) and Jian Kang (Inst. for Environ. Design and Eng., Univ. College London, London, United Kingdom)

The field of soundscape studies considers sound environments as perceived, in context, and has recently focussed on understanding the potential of physical acoustical features, including temporal characteristics, for predicting human perception. This study investigates the presence of a 1/f structure in the power spectrum slope of six (psycho)acoustical parameters of 30-s recordings of urban acoustic environments. The acoustical parameters were calculated as time series throughout the recording period, then the power spectrum of the time series was calculated and plotted on a log-log scale, with the x-axis ranging from 30 s/$10^{-1.5}$ Hz to 0.01 s/10^{2} Hz. The slope of the best-fitted straight line through the power spectrum was calculated and compared to the corresponding perceptual attribute ratings of the soundscape collected on site during the recording. The dataset includes 300 + recording-response pairs. An overall 1/f structure was not found for any of the parameters. Differences in temporal behaviour are indicated at different time scales, with a deviation in slope typically occurring at ∼2 s/0.5 Hz, reflecting differences in temporal behaviour among within-sound-event time scales and between-sound-event time scales. An ordinal logistic regression model is developed which predicts the perceptual attributes of urban soundscapes based on the spectral slopes of the acoustic time series.

4:00

1pSP9. Application of lp-norm regularisation techniques in the synthesis of indoor tyre pass-by noise with the inverse method. Athanasios Papaioannou (Inst. of Sound & Vib. Res., Univ. of Southampton, University Rd., Southampton SO17 1BJ, United Kingdom, A.Papaioannou@soton.ac.uk), Stephen Elliott, and Jordan Cheer (Inst. of Sound & Vib. Res., Univ. of Southampton, Southampton, Hampshire, United Kingdom)

The indoor pass-by noise measurement can nowadays be realised in a laboratory environment with a far field microphone array and a stationary vehicle on a rolling road, according to ISO-362-1:2016. Within this indoor testing procedure, there exists the ability to quantify the contributions from the various noise sources in a car to gain further insight into the total noise disturbance. The tyre noise contribution can be estimated by using an additional set of microphones close to the tyre and performing an inverse method. This work assumes measured near field pressure data close to a rolling tyre at different car speeds. The inverse method is utilised to find the corresponding source strength estimates which are then used to synthesise the far field tyre noise contribution and compare it with the one measured directly at the far field microphone array. A combination of novel l1 regularisation and conventional l2 regularisation methods is finally investigated to further optimise the equivalent source strength estimates and, subsequently, the far field pass-by noise pressure estimates through the frequency range of interest.

4:20

1pSP10. An efficient DSP network for the real-time auralization of complex urban scenarios. Michael Vorlaender (ITA, RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52056, Germany, mvo@akustik.rwth-aachen.de), Henry Andrew, and Jonas Stienen (ITA, RWTH Aachen Univ., Aachen, NRW, Germany)

The most commonly used technique for auralization is a simulation step to determine sound propagation based on Geometrical Acoustics and a subsequent assembly of the simulation results into a filter or (binaural) impulse response. Convolution is used to apply the propagation effects to an anechoic recording per source-receiver pair, which is efficient for a scenario including a few sound sources and many propagation paths. The procedure offers a good balance of speed and accuracy, and has seen much development. However, it reveals an intrinsic difficulty upon scene adaption (filter exchange) for fast-moving sound sources, because in principle it requires a steady-state environment. In particular, this constellation is found in urban scenarios, where many fast-moving sound sources can be expected while essentially only a manageable number of propagation paths contribute to the perceived overall sound field. An efficient DSP network is put to discussion that handles propagation paths individually using a single-input multiple-output variable delay line per sound source and applies spectral influences like diffraction effects using short IIR filters. Binaural cues are maintained by a clustering approach that combines incident waves and offers a constant convolution cost.

Session 1pUWa

Underwater Acoustics: Scattering and Reflection

Sean Pecknold, Chair
DRDC Atlantic, PO Box 1012, Dartmouth, Nova Scotia B2Y 3Z7, Canada

Contributed Papers

1:00

1pUWa1. Line-scan imaging of monostatic scattering by a sphere near a flat interface: Identification of direct, indirect, and multiple-scattering paths. Auberry R. Fortuner (Phys., WSU, Webster Hall 1245 PO Box 2814, Pullman, WA 99164-2814, auberry.fortuner@wsu.edu) and Philip L. Marston (Phys., WSU, Pullman, WA)

Quasi-holographic vertical scans of monostatic scattering give an image in cross-range and time that separates and focuses the different scattering contributions. The scattering of a sphere near a flatinterface is modified by reflections at the interface that are focused in the image at locations depending on the target depth and size. Labroratory experiment was performed in which the monostatic scattering from a soft spherical target near a free-surface was recorded along a vertical line scan. A geometric model describes the contributing paths. The 1st order single-scattering and 2nd and higher-order multiple-scatterings are identified by their location in the quasi-holographic image. The reversibility of the image processing allows individual echoes to be isolated, and this was done to compare in the time and frequency domain the direct specular reflection with the multiple-scattering echoes. The experiment was repeated for various target depths and the dependence of multiple-scattering echo amplitudes with target depth was examined. Finally, a comparison is made between this case and that of scattering from two adjacent targets from a similar line scan and the resulting quasi-holographic image examined elsewhere [D. S. Plotnick and P. L. Marston, in EUSAR 2016 Proceedings]. [Work supported by the Office of Naval Research.]

1:15

1pUWa2. Measured surface and bottom reflection loss and scattering on the Chukchi Shelf observed during the 2016–2017 Canada Basin Acoustic Propagation Experiment. Sean Pecknold (DRDC Atlantic, PO Box 1012, Dartmouth, NS B2Y 3Z7, Canada, sean.pecknold@drdc-rddc.gc.ca), Carolyn M. Binder (DRDC Atlantic, Dartmouth, NS, Canada), Mohsen Badiey (Univ. of Delaware, Newark, DE), Jason D. Sagers, Megan S. Ballard (Environ. Sci. Lab., Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Altan Turgut (Acoust., Naval Res. Lab., Washington, DC)

During the 2016–2017 Canada Basin Acoustic Propagation Experiment (CANAPE), acoustic transmissions were recorded from both deep- and shallow-water sources on a set of moored receivers, over the course of nearly one year, in ice-covered and open water conditions. One of the source-receiver geometries included LFM transmissions of 4000–1500 Hz and 1100–700 Hz recorded on an eight-element volumetric array at a distance of approximately 400 m. Surface, bottom, and sub-bottom reflections are observed in the data and are used to infer ice and sediment geo-acoustic properties. Some observations about bistatic scattering are also made.

1:30

1pUWa3. Using the frequency-difference autoproduct to regain coherence in rough surface scattering. Nicholas J. Joslyn (Appl. Phys., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, njoslyn@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Incident and reflected acoustic waves, with wavenumber k, are coherent in an ideal Lloyd's mirror environment. However, a random rough surface, characterized by its rms roughness height h, reduces this coherence as kh increases. This presentation describes the ability of the frequency-difference autoproduct to recover reflected-field coherence, albeit at a lower frequency, even in the presence of high kh values. The frequency-difference autoproduct is a quadratic product of complex field amplitudes at different frequencies within the signal bandwidth. Prior work has shown that the frequency-difference autoproduct can mimic a below-band field at the difference frequency. Thus, by choosing a sufficiently-low difference frequency, the apparent surface roughness can be reduced so that a rough-surface scattered field resembles a flat-surface reflected one. For the theory and simulation results presented here, the surface roughness is Gaussian distributed and isotropic, and the Kirchhoff (tangent plane) approximation is used to determine the reflected sound. For $1 < kh < 6$, theoretical and numerically simulated autoproducts in a rough-surface environment are compared to acoustic fields at the difference frequency in an ideal Lloyd's mirror environment. Accompanying experimental results may be provided as well, if available. [Work supported by ONR.]

1:45

1pUWa4. Interference structure in the form function for bistatic scattering from elastic spherical shells in water. Auberry R. Fortuner (Phys., WSU, Webster Hall 1245 PO Box 2814, Pullman, WA 99164-2814, auberry.fortuner@wsu.edu) and Philip L. Marston (Phys., WSU, Pullman, WA)

Although the monostatic scattering from a sphere is independent of viewing angle, the bistatic frequency response contains rich structures a function of scattering angle. The exact solution for the far-field scattering from underwater elastic spherical shells is examined as a function of scattering angle. The high-frequency scattering from spherical shells includes contributions from leaky Lamb-type waves that circumnavigate the shell while continuously radiating into the fluid. The form function in scattering angle contains multiple interference patterns between the specular reflection and the different modes of Lamb waves. This interference structure is similar to the case for solid spheres and cylinders recently studied [A.M. Gunderson, A.L. España, and P.L. Marston, *J. Acoust. Soc. Am.* **142**, 110–115 (2017)], where the interference patterns between the specular reflection and Rayleigh waves were examined. For the elastic shell case, the greater number of Lamb wavemodes and their highly dispersive nature creates richer interference structure in the form function from that of the solid sphere. The interference loci between the specular reflection and the two lowest mode Lamb waves are calculated and compared with the bistatic form function. [Work supported by the Office of Naval Research.]

2:00

1pUWa5. High-frequency backscattering enhancements from elastic cylindrical shells in water: Observations and ray theory. Bernard R. Hall (Dept. of Phys. and Astronomy, Washington State Univ., 100 Dairy Rd., Rm. 1245, Webster Hall, Pullman, WA 99164, bernard.hall@wsu.edu) and Philip L. Marston (Phys., Washington State Univ., Pullman, WA)

Sometimes elastic features for acoustically illuminated targets in water are relegated primarily to low frequency bands. For empty spherical shells however, a pronounced high-frequency backscattering elastic-enhancement was observed and modeled using ray-theory [G. Kaduchak, D. H. Hughes, and P. L. Marston, *J. Acoust. Soc. Am.* **96**, 3704–3714 (1994)]. That phenomenon was termed a backwards wave enhancement because the wave phase and group velocities guided on the shell move in opposite directions and required a modified ray diagram. In the present research, this type of backscattering enhancement is demonstrated and modeled for empty cylindrical elastic shells in water. While the full ray-theory requires solving for complex roots descriptive of elastic waves on fluid-loaded shells, it was found helpful to first consider the properties of ordinary Lamb waves on plates whose thickness was selected to be that of the shell of interest. A search for different roots of Lamb's Equation confirms that a root exists that exhibits opposing phase and group velocities in the frequency region of interest. As expected from the model, the associated backscattering enhancements are localized in frequency though they can be easily identified using tone bursts. [Work supported by Office of Naval Research.]

2:15

1pUWa6. Limited duration integral equation method. Edward Richards (Scripps Oceanogr., Univ. of California, San Diego, 8820 Shellback Way, Mail Code 0238, La Jolla, CA 92093, edwardlrichards@ucsd.edu), Hee-Chun Song, and William S. Hodgkiss (Scripps Oceanogr., Univ. of California, San Diego, La Jolla, CA)

This talk describes the limited duration integral equation method (DIEM), a reference solution for time domain scatter from one-dimensional surfaces. The DIEM approximates the exact Helmholtz integral equation by truncating this spatially infinite integral. The DIEM truncation causes error in its solution, which is mitigated using arrival time predictions from the analytically simpler Helmholtz-Kirchhoff approximation (HKA). The HKA is additionally shown to effectively replace much of the DIEM convergence analysis, leading to efficient DIEM solutions. The exact Rayleigh-Fourier method (RFM) is used to validate the DIEM result for a sinusoidal scattering surface scenario. While the HKA yields significant error in this scenario, it predicts the truncation behaviors of the DIEM correctly. Unlike the RFM that is specific to periodic surfaces, however, the DIEM is not restricted to any specific surface geometry. The DIEM therefore proposed as an efficient reference solution for future line source scattering studies with general scattering surfaces.

2:30

1pUWa7. Study on phase of reflection coefficient of the medium with gradient change of acoustic impedance. Bo Hu (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin Eng. University, Harbin, Heilongjiang 150001, China, kidd1105@sina.com)

Medium with gradient change of acoustic impedance is a new type acoustic structure which composed of several materials and has a continuous gradient composition and structure whose specific acoustic impedance varies smoothly across the layer. It has been used in various practical situations such as acoustic rectifiers, medical devices, weakening vibration and reducing noise. The purpose of this paper is to investigate the phase characteristics of reflection coefficient of medium with gradient change of impedance to make use of the sound pressure and vibration velocity of the wave. Exact solutions of the Helmholtz equation are derived for a typical density profile and sound-speed profile. The solutions in the medium are matched with solutions in the homogeneous upper and lower layers to derive analytical expressions for the phase of the reflection coefficient of a plane wave which incident from the upper medium. The phase of reflection coefficient in a variety of cases that of different frequencies and incident angles are examined. A number of simulation results are presented and comparison is made with computed results. The phase data can be used in the subsequent vector signal processing.

Session 1pUWb

Underwater Acoustics: Detection, Classification, and Noise

Shane Lani, Cochair
Johns Hopkins Applied Physics Lab., Laurel, Maryland 20723

Daniel J. Brooker, Cochair
Underwater Acoustics, Navy Research Lab, Washington, DC 20375

Contributed Papers

3:15

1pUWb1. Experimental analysis of the effects of bistatic target scattering on synthetic aperture sonar imagery. Thomas E. Blanford (Penn State Univ., Penn State Univ., State College, PA 16804, teb217@psu.edu), Shawn Johnson (Penn State Univ., State College, PA), and Daniel C. Brown (Penn State Univ., State College, PA)

Downward looking synthetic aperture sonar (SAS) systems, such as those used to detect surficial and buried unexploded ordinance, may use arrays with large spatial extent compared to the depth of the targets they are imaging. With such systems the targets are in the nearfield of the physical array formed by the projector and receiver. Beamforming algorithms, data representation schemes, and automated target recognition algorithms can benefit from considering the bistatic scattering patterns of targets in this geometry. This presentation will describe experimental analysis of the effects on SAS imagery due to bistatic scattering from cylindrical targets. A set of field measurements of elastic and inelastic acoustic scattering from open and closed ended cylinders was collected in a lake environment in less than 3 m of water. These data will be compared to analogous, in-air laboratory measurements. Finally, possible methods for automatically detecting elastic scattering behavior from bistatic collection geometries will be discussed.

3:30

1pUWb2. Passive detection on a distributed network using information geometry. Daniel J. Brooker (Underwater Acoust., Navy Res. Lab, 3201 Landover St. Apt. 1504, Alexandria, VA 22305, daniel.brooker@nrl.navy.mil), Steven I. Finette, and Peter C. Mignerey (Underwater Acoust., Navy Res. Lab, Washington, DC)

Using data from the recent "Distributed Network Consensus" (DSNCON19) experiment several aspects of information geometric source detection are tested. In this experiment, acoustic data were collected on seven spatially separated bottom mounted arrays in a "truss" configuration approximately 60 NM off the coast of New Jersey using towed and autonomous CW sources. Recent proposals for new detectors based on the averaging of covariances matrices from a single array according to principles of Riemannian geometry are tested against conventional passive detection schemes. In addition, new information-geometric passive detectors are developed for the distributed network of arrays. The possibility of source localization using the network is also considered. [Work supported by the Office of Naval Research.]

3:45

1pUWb3. Comparison of sparse Bayesian learning and atomic norm minimization in detecting passive tonal signals. Myoungin Shin (Sejong Univ., Neongdong-ro 209, Gwanggaetogwan 702ho, Gwangjin-Gu 05006, South Korea, myoungin@sju.ac.kr), Youngmin Choo, Keunhwa Lee, and Wooyoung Hong (Sejong Univ., Seoul, South Korea)

The frequency estimation of tonal signals in passive sonar systems is crucial to the identification of the marine object. In this study, we introduce the frequency estimation of tonal signals using the multiple measurement vector atomic norm minimization(MMV-ANM) technique based on compressive sensing (CS) and sparse Bayesian learning (SBL) technique based on Bayesian Inference. CS techniques have been employed in identifying the tonal frequency components. It is now well-known from the theory of CS that a high-dimensional signal can be recovered from a relatively small number of measurements as long as the desired signal is sparsely represented. MMV-ANM, an extended version of CS technique for solving the basis mismatch problem, has been proposed and shown better performance in terms of resolution and stability. On the other hand, tonal signals at passive sonar systems are also estimated with sparse Bayesian learning (SBL) using MMV. The prior source amplitudes are assumed independent zero-mean complex Gaussian distribution with the unknown variances (hyperparameters). The hyperparameters are derived by maximizing the probability of given passive data, which leads to sparse tonal signal frequencies. We compare performance of SBL and MMV-ANM in estimation of passive tonal signals with synthetic and *in situ* data. [*corresponding author Youngmin Choo]

4:00

1pUWb4. A method for undersea gas bubbles detection from acoustic image. Wanyuan Zhang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No.145, Nantong St., Nangang Dist., Harbin 150001, China, zhangwanyuan@hrbeu.edu.cn), Yajun Wang, and Tian Zhou (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China)

In recent years, there has been an increasing requirement for methods of detecting bubbles released from the seabed into the water column, such as leaks from undersea gas pipelines and seeps from carbon capture and storage facilities. Considering the peculiarity of the layout of submarine gas pipelines and that of the ocean environment around them, we construct an underwater mobile platform equipped with autonomous underwater vehicles carrying multi-beam sonars and various types of other sensors. Analogous to optical flow, this paper describes a scale-invariant feature transform (SIFT) flow algorithm, which includes both the detection of key points and the computation of local descriptors. This SIFT flow algorithm estimates the motion characteristics of gas leaks and quantifies the flux of gas. Finally, the validity of this method is verified in pool and sea experimental research.

4:15

1pUWb5. Sparsity-based frequency-domain adaptive line enhancer. Guolong Liang (Acoust. Sci. and Technol. Lab., Harbin Eng. Univ., Harbin, Heilongjiang, China), Yu Hao (Key Lab. of Marine Information Acquisition and Security (Harbin Eng. University), Ministry of Industry and Information Technol. 145 Nantong St., Nangang Dist, Harbin, Heilongjiang 150000, China, haoyu@hrbeu.edu.cn), Nan Zou (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China), and Longhao Qiu (Qingdao Haina Underwater Information Technol. Co., Ltd., Harbin, China)

The radiated lines from underwater targets are an important feature for passive sonar detection. Adaptive line enhancer (ALE) is usually applied as a preprocessing step to enhance the signal-to-noise ratio (SNR) of the lines. However, the conventional ALE based on least-mean-square (LMS) algorithm suffers from the weight noise in the adaption, which limits the SNR gain severely. Inspired by the frequency-domain sparsity of the lines, a sparsity-based ALE is developed to break through this limit. The proposed ALE is implemented in the frequency domain and a sparse penalty is incorporated into the frequency-domain adaption. By means of the sparse penalty, the weight noise is suppressed and the SNR gain is well improved. Simulation results demonstrate that the SNR gain of the proposed ALE is 9 dB higher than that of the conventional ALE. Experimental data processing also verifies the superiority of the proposed ALE.

4:30

1pUWb6. Examination of active target detector using *in situ* data. Youngmin Choo (Defense System Eng., Sejong Univ., 209 Neungdong-ro, Gwangjin-gu, Seoul 05006, South Korea, ychoo@sejong.ac.kr), Mingu Kang, Wooyoung Hong, and Keunhwa Lee (Defense System Eng., Sejong Univ., Seoul, South Korea)

When operating active sonar systems, target-like signals including clutter are detrimental to detecting target signals and induce an increment of false alarms. In this work, various approaches including machine learning and deep learning techniques are applied to in-situ data to investigate their performance. First, one of conventional schemes, constant false alarm rate (CFAR) detector is used to find target signals in acoustic data. It detects target signals with high SNR, but it also captures high-intensity clutter signals, which leads to many false alarms. Subsequently, data-driven decision rules from machine learning of support vector machine (SVM) and deep learning of convolutional neural network (CNN) are applied. Both of SVM using aural features and CNN with spectrogram show better accuracy. In particular, false alarms from the CFAR detector are extremely decreased owing to the data-driven decision rules not exploiting the signal intensity for target detection.

4:45

1pUWb7. Low to mid frequency whale noise compared to ambient noise. Shane W. Lani (Johns Hopkins Appl. Phys. Lab., 1454 Catherine St., Decatur, GA 30030, lani.shane@gatech.edu)

While the majority of ocean noise in the low and mid frequency range originates from the surface (shipping, wind wave noise), biologics are the predominant source of noise originating below the surface. Of the biologics, whales are one of the loudest and most frequent producers of vocalizations in the LF and MF bands. Their songs and clicks can be as loud as 236 dB re 1 μPa at 1 m and range in frequency from 10 Hz to well above 10 kHz. For a given frequency band, the ambient noise (excluding biologics) is typically greater than the cumulative effect of all contributing whales; however, as larger hydrophone arrays are built with increasing amounts of array gain this is no longer always the case. This work explores various spatial densities and source pressure levels (SPL) of whales to determine when a collection of whales becomes the dominant noise source on an array.

5:00

1pUWb8. Extensive underwater radiated noise analysis of a research vessel in compliance with DNV SILENT notation. Ruey-Chang Wei (Inst. of Undersea Technol., National Sun Yat-sen Univ., 70 Lienhai Rd., Kaohsiung City 804, Taiwan, rcwei@mail.nsysu.edu.tw), Chi-Chun Wang, Pai-Ho Chiu (Inst. of Undersea Technol., National Sun Yat-sen Univ., Kaohsiung City, Taiwan), and Bo-Jhih Huang (CSBC Corp. Taiwan, Kaohsiung City, Taiwan)

Studies have shown that underwater-radiated noise from ships may have both short and long-term negative consequences on marine life, especially marine mammals. An international community of researchers, environmental groups, government agencies, and sectors of the shipping industry has recognized shipping noise as an important marine conservation issue. In 2010, Det Norske Veritas (DNV, the Norwegian Classification Society) made the first attempt to fix the radiated noise limits for commercial ships, in which vessels fulfilling the requirements will be given an optional class notation SILENT. In this study, underwater noise measurements were performed on a new research vessel in shallow water in compliance with DNV SILENT notation. More speeds were tested than required by DNV, to understand the characteristics of related noises. Results show the new ship satisfies the allowable noise level curve for type R (Research); however, the measured noise levels do not increase as speed accordingly. Moreover, level differences can be significant between starboard and port side tracks especially at low frequencies. Commonly observed coastal currents play an import role on ship maneuverings and test conditions and therefore unexpected results. Practical considerations and suggestions about performing measurements and analyzing the data will be discussed and summarized.

5:15

1pUWb9. Acoustic behavior of snapping shrimp in east china sea. DaeHyeok Lee (Dept. of Marine Sci. and Convergence Eng., Hanyang Univ., 55, Hanyangdaehak-ro, Sangnok-gu, Ansan-si, Gyeonggi-do 15588, South Korea, edh0921@hanyang.ac.kr), Myungkwon Ko (Dept. of Marine Sci. and Convergence Eng., Hanyang Univ., Ansan, Gyeonggi do, South Korea), and Jee Woong Choi (Dept. of Marine Sci. and Convergence Eng., Hanyang Univ., Ansan, South Korea)

Ambient noise in shallow water is typically more complicated than that in deep sea. The snapping shrimp, which is a major source of ambient noise in shallow water, produces impulsive-shape noise. During the period of 14–28 May 2015, the Shallow-water Acoustic Variability Experiment (SAVEX15) was performed in the East China Sea (ECS) at a shallow water off the southwest of Jeju Island, Korea. In the study area, snapping shrimp noise was recorded continuously with dominant frequencies higher than 3 kHz. Envelope correlation method combined with the threshold detection was used to extract the snapping events from the received ambient noise data. Although many previous research studies have reported that the diurnal variation of snapping shrimp noise is mainly due to water temperature variation controlled by light availability, the observed snap rates in our experimental site showed unusual sinusoidal pattern and seemed to be positively correlated with the phase of the tidal current. [Work supported by the Development of Civil Military Technology Program (No. 18-SN-RB-01) from the Institute of Civil Military Technology Cooperation (ICMTC) of the Republic of Korea.]

5:30

1pUWb10. Time-frequency characteristics estimation of underwater moving objects using memory-dependent derivative method. Weitao Sun (Northwestern PolyTech. Univ., 127 West Youyi Rd., Beilin Dist., Xi'an, Shaanxi 710072, China, sunwt1223@Gmail.com), Huigang Wang, Qingyue Gu, Yifeng Xu, and Shaowei Rong (Northwestern PolyTech. Univ., Xi'an, Shaanxi, China)

The line-spectra changes of the radiated noise of underwater moving object is observed as the form of narrow-band time varying signal due to the Doppler effect. The modulation law of the time varying signal contains a

large number of feature information of moving targets, which can be used for detection and classification. The goal of time-frequency analysis is to extract subtle changes the function relationship of frequency over time. In this research, a memory-dependent derivative methodology is proposed to deal with accurate time–frequency representation of time varying signals under strong background noise. Memory-dependent derivative is a convolution of a common derivative by the time-varying signals with a dynamic weighted function in the past period time. Considering the stated methodology, it is derived that the discrete data from previous times to estimate signal value at current time and reduce the effects of the noise. Using the Fourier transformation with different scales and delays transformation as the kernel function, the energy concentrated time-frequency curve is obtained with higher resolution and without frequency leakage. The simulated results demonstrate that given method is immune to background noise and estimate the main frequencies with high accuracy, specially the rapid change of the modulated frequency.

MONDAY AFTERNOON, 2 DECEMBER 2019 REGENT, 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Effective Media Interactions Training Workshop

Laura Kloepper, Cochair
Biology, Saint Mary's College, 262 Science Hall, Saint Mary's College, Notre Dame, Indiana 46556

Andrew A. Piacsek, Cochair
Physics, Central Washington University, 400 E. University Way, Ellensburg, Washington 93407

The Public Relations Committee and the AIP Media Services team present this hands-on workshop for meeting attendees who are interested in effectively communicating scientific work to the public. This workshop is strongly recommended for individuals who regularly speak to reporters or who may do so in the future. The workshop will consist of short presentations by media professionals to provide a toolkit of specific ideas and techniques for speaking to the media as well as structured small group activities that will give participants an opportunity to discuss and apply those techniques. Participants should come prepared to give a one-minute "elevator talk" about their own research. The workshop will provide some perspective of the "other side"—how journalists and other media professionals approach their work—and participants will leave with a strong understanding of best practices, recommendations and strategies for ensuring their scientific ideas are correctly interpreted.

Exhibit

An instrument and equipment exhibition will be located in the Ballroom near the registration area and meeting rooms.

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

Monday, 2 December, 5:30 p.m. to 7:00 p.m.: Exhibit Opening Reception including lite snacks and a complimentary drink.

Tuesday, 3 December, 9:00 a.m. to 5:00 p.m.: Exhibit open hours including a.m. and p.m. breaks serving coffee and soft drinks.

Wednesday: 4 December, 9:00 a.m. to 12:00 noon: Exhibit open hours including an a.m. break serving coffee.

Session 2aAA

Architectural Acoustics and Computational Acoustics: Computational Acoustics for Architectural Applications

Laura C. Brill, Cochair
Threshold Acoustics, 141 W Jackson Blvd, Suite 2080, Chicago, Illinois 60607

Michael Vorlaender, Cochair
ITA, RWTH Aachen University, Kopernikusstr. 5, Aachen 52056, Germany

Chair's Introduction—8:00

Invited Papers

8:05

2aAA1. Novel applications of re-tooled open-source acoustic simulation algorithms. Arthur W. van der Harten (Pachyderm Acoust. Sci., 145 Huguenot St., New York City, NY 10801, Arthur.vanderharten@gmail.com) and Cameron Goodman (Acoust. Distinctions, Brooklyn, NY)

Pachyderm Acoustic Simulation is an open-source acoustics simulation tool that is freely available to the public through the General Public License 3.0. In the 11 years, that Pachyderm Acoustic Simulation has been available, its open and conventional acoustics simulation algorithms (including geometrical, finite volume, transfer matrix, etc.) have been applied in ways that are less conventional by certain creative individuals. Users leverage access to the algorithms through its native interface, grasshopper (for visual generation of algorithms), IronPython, and the software's native source language C#. Examples of these creative applications include augmented geometrical visualizations, customized finite volume method simulations for proof of concept, visualization of material data, and form-finding using genetic algorithms. This talk will explore several exemplary cases in which a user successfully applied the simulation and calculation algorithms in a way that was not originally intended.

8:25

2aAA2. Interactive rendering of dynamic virtual audio-visual environments for subject-in-the-loop experiments. Giso Grimm, Maartje M. Hendrikse (Medical Phys. and Acoust., Cluster of Excellence Hearing4all, Univ. of Oldenburg, Oldenburg, Germany), and Volker Hohmann (Medical Phys. and Acoust., Cluster of Excellence Hearing4all, Univ. of Oldenburg, Postfach, Oldenburg 26111, Germany, volker.hohmann@uni-oldenburg.de)

The benefit from hearing devices with directional filtering algorithms, e.g., hearing aids with fixed beamformers, may depend on the user's head movement behavior. Traditional evaluation methods for hearing devices often do not allow for head movements, or present stimuli in tasks which do not result in natural movement behavior of the participants. To overcome these limitations, virtual audio-visual environments are increasingly used to reproduce listening environments. Here, we present a software tool for interactive real-time simulation of audio-visual virtual environments. Simulation methods for time-domain rendering of direct sounds and early reflections in a geometric image source model in combination with a simple method of rendering diffuse sound fields from first-order Ambisonics recordings as well as diffuse reverberation from recorded first-order Ambisonics impulse responses are presented. A set of five typical and difficult listening environments was created in the lab with loudspeaker reproduction, and the movement behavior of 40 young and elderly normal hearing listeners was recorded in these environments. Furthermore, the influence of this individual head movement data on the performance of various signal enhancement algorithms was simulated. Data show that the application of virtual acoustic environments may contribute to a higher ecological validity of lab-based hearing-device evaluation methods.

8:45

2aAA3. Binaural models that can assess and understand architectural environments. Jonas Braasch (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu) and Jens Blauert (Institut für Kommunikationsakustik, Ruhr-Universität Bochum, Bochum, Germany)

Traditional binaural models, such as those based on cross-correlation and equalization/cancellation algorithms, use a bottom-up approach. They simulate the functions of the ascending auditory pathway from signals at the ears up through higher-level structures, typically the midbrain. In contrast, top-down methods are frequently used in practical robotics applications, for instance, when applying simultaneous-localization-and-mapping (SLAM) techniques. Such top-down approaches, however, are generally absent from models that foster understanding of our auditory system. This talk focuses on binaural-localization models that integrate traditional signal-driven models with psychoacoustically motivated top-down structures, for example, by means of feedback loops. The aim is to design advanced

models that operate in architectural environments in order to understand both the environment and activities within the environment. These models are presented in the context of the authors' current work on a contributed book entitled, "*The Technology of Binaural Understanding*," which is in press at Springer. The discussed approaches include: models to estimate room-impulse responses and binaural-activity maps from running signals; algorithms to utilize head movements for strategically reading the environment; and algorithms for the automatic labeling of sonic events using deep-learning techniques. [Work supported by the EU-FET project "TwoEars," Contract No. 618075, NSF BCS-1539276, and CISL (RPI).]

9:05

2aAA4. Acoustic simulation in an orchestra rehearsal room using BRASS software. Guilherme C. Fagerlande (Faculty of Architecture and Urbanism, Federal Univ. of Rio de Janeiro, Cidade Universitária, Av. Reitor Pedro Calmon, 550, Sala 433, Rio de Janeiro 21941-590, Brazil, guilhermefagerlande@gmail.com), Julio Cesar B. Torres (Polytechnic School, Federal Univ. of Rio de Janeiro, Cidade Universitária, Rio de Janeiro, Brazil), and Maria Lygia A. Niemeyer (Faculty of Architecture and Urbanism, Federal Univ. of Rio de Janeiro, Cidade Universitária, Brazil)

The Music School of the Federal University of Rio de Janeiro is located in a heritage-listed building, due to the historical and architectural relevance. Acoustic problems were related to the musicians in the orchestra rehearsal room, such as high reverberation time on certain frequencies and insufficient insulation. The aim of this paper is to analyse the space of the rehearsal room through measurements and to propose corrections through simulations using BRASS (Brazilian Room Acoustic Simulator), a software designed by the Polytechnic School of UFRJ, to achieve acoustic adequacy. The methods used are measurements of mono and binaural impulsive responses in various positions, computational modeling, simulation verification by comparison with acoustic parameters (T_{60}, EDT, CT, C_{80}, D_{50}, etc), and proposition of architectural corrections and improvements in the room with subjective evaluation by auralization for the musicians. The results show the importance of the acoustic simulation method in the design process of music rehearsal rooms, in order to obtain efficient solutions for the use, besides being able to reproduce the rooḿs sound with auralization, and respecting its original architectonical characteristics.

9:25

2aAA5. Overhead stage canopies in a coupled volume theatre: Effects on the sound energy distribution and on the secondary reverberation. Dario D'Orazio (DIN, Univ. of Bologna, Viale Risorgimento, 2, Bologna 40128, Italy, dario.dorazio@unibo.it), Giulia Fratoni, and Massimo Garai (DIN, Univ. of Bologna, Bologna, Italy)

Nowadays, opera houses are often used for symphonic music, even though the intrinsic characteristics of these theatres are not suited for this purpose, due to their coupled volumes and high absorption of the fly tower. When symphonic music is performed in these halls an overhead stage canopy is often used to enhance the orchestral performance. In the present work, the effects of a canopy array in a coupled volume theatre were studied. The array canopy was designed and installed based on Geometric Acoustic (GA) simulations calibrated with *in-situ* measurements. Results showed peculiar effects on the sound energy distribution through space: the sound strength values depends on the "effective" volume of the theatre, varying with the sound source position. Moreover, when the stage is covered by the canopy array, the sound strength depends on the distance from the aperture instead of the distance from the sound source position. In other words, the decay curve is "tilt" by "effective volume" and "shifted" by the canopy array. Furthermore, the changes in sound behaviour due to the canopy array may be considered as a switch-off of the secondary reverberation effect.

9:45

2aAA6. An efficient hybrid VBAP—Ambisonics convolution technique for real time auralization. Matthew Azevedo (Acentech, Inc., 33 Moulton St., Cambridge, MA 02138, mazevedo@acentech.com)

Ambisonics requires high orders with many channels for high spatial accuracy, which makes real-time convolution of many HOA (high order ambisonics) sources computationally challenging. We have developed a technique to process an ambisonic impulse response to isolate the direct and early components and spatialize them with vector-based amplitude panning and using simple 2nd order ambisonics to present the diffuse late energy. This hybrid approach results in sharp imaging of the auditory scene while only requiring 14 channels of convolution per source.

10:05–10:20 Break

10:20

2aAA7. Computations of sound power and level components in symphony orchestra. Magne Skalevik (AKUTEK and Brekke&-Strand, Bolstadtunet 7, Spikkestad 3430, Norway, msk@brekkestrand.no)

A symphony orchestra and its members are both sources and receivers. The individual musician in an orchestra hears a combination of direct sound from own instrument, a mix of direct sounds from co-musicians, and reverberant sound from the whole orchestra. The delicate balance between these three components is assumed to have big impact on playing conditions, hearing of self and others, masking of self and others, ensemble, self-reinforcing feedback loops that cause escalating sound power, and excessive noise exposure, to mention a few. Of course the result will affect the audience. Different models for computing the three sound components, and results will be presented.

10:40

2aAA8. Application of Hilbert space operators on the sphere to directivity measurements. Samuel Bellows (Dept. of Phys., Brigham Young Univ., Provo, UT 84602, samuel.bellows11@gmail.com) and Timothy W. Leishman (Brigham Young Univ., Provo, UT)

Spherical directivity measurements are useful for understanding and characterizing acoustic sources and are often applied to computer-based modeling and simulation of acoustic spaces. While the measurements must be handled with great care for suitable accuracy, little if any data post-processing has been employed in the past to ameliorate them and improve their utility. This paper outlines techniques utilizing Hilbert space operators on the unit sphere to both evaluate certain qualities of directivity measurements and provide refinements of their datasets. Truncation, projection, and symmetry operators on directivity measurements are specifically shown to benefit post-processing of the spherical functions.

10:55

2aAA9. An archival database of high-resolution directivities. Rachel C. Edelman (Phys. and Astronomy, Brigham Young Univ., 475 w 1720 n, Apt 1-303, Provo, UT 84604, rachel.edelman17@gmail.com), Samuel Bellows, and Timothy W. Leishman (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Having easily accessible and accurate directivity patterns of sound sources is valuable for many applications, including architectural acoustics modeling and spatial audio. To provide this information to other researchers, the authors of this presentation are creating an online directivity database for live human speech, played musical instruments, and other sources of sound. The results are derived from recordings of the sources over their useful bandwidths at 2522 unique microphone positions over a surrounding sphere (i.e., with 5-deg resolution in both the polar and azimuthal angles). Processing of the recordings has led to frequency-dependent spherical-harmonic expansions. The expansion coefficients, as well as broad-band tabulated attenuation results (commonly used in architectural acoustics simulation packages), are freely available in the ASCII format. The database also contains figures and animations of the directivity patterns, allowing for quick visualization. The collections should help improve modeling of various acoustic spaces, microphone placements for recordings, and general understanding of source radiation characteristics.

11:10

2aAA10. Developing a new method for analyzing room acoustics based on auralization. Alaa Algargoosh (Architecture, Univ. of Michigan, 2000 Bonisteel Blvd, Ann Arbor, MI 48109, alaas@umich.edu) and John Granzow (Performing Arts and Technol., Univ. of Michigan, Ann Arbor, MI)

Room modes are the resonances that result from generating a sound in a room. The sound coloration produced by these modes is often deemed problematic. Research in room acoustics continues to refine our means to control this coloration for reconciling visual and aural variables in architectural design. Alvin Lucier demonstrated the resonant frequencies of a room in his sound art piece “I am sitting in a room” by recording his speech in a space then playing the recording and re-recording it in the same room multiple times until the resonant frequencies of the room became dominant. Inspired by Lucier’s work, this research explores the possibilities that can emerge from replicating the same iterative process within a simulated framework. Accordingly, a room is modeled, an impulse response (IR) is generated and an auralization is created by convolving an anechoic recording with the IR. The output is then used as an input that is convolved again with the same IR. This method is similar to the method used in artificial reverberation to control the feedback loops of PA systems and is expected to advance room acoustic analysis. This paper discusses the benefits and limitations of the proposed method with possible applications.

11:25

2aAA11. Prediction of sound fields after propagation through sound barriers by CNN and DCNN algorithms. Jonghwan Kim (Hanyang Univ., Seoul KS013, South Korea, givemeletter@hanmail.net), Gwanghoon Jung, and Junhong Park (Hanyang Univ., Seoul, South Korea)

Sound fields are affected by the size of the space and the surrounding environments. The ray tracing method analyze the sound propagation through geometric constraints. For prediction of sound fields in a real-time to investigate the influence of surrounding environments, an effective calculation method is required. During the prediction, it is necessary to calculate the sound field distribution according to the layout of equipment or sound reflecting devices to minimize noise in working spaces. This procedure is required for an optimal arrangement of barriers according to noise sources. To improve this repetitive work, it is devised to develop a machine learning model that predicts the change in the sound field according to the position of sound barrier through CNN and DCNN algorithms. Training data were generated by using a commercial tool. The proposed approach recognizes the characteristics of sound fields at spaces after acoustic barriers.

Session 2aAB

Animal Bioacoustics and Acoustical Oceanography: Low-Frequency Sound Production and Passive Acoustic Monitoring I

Jack Butler, Cochair
Marine Physical Lab, Scripps Institution of Oceanography, La Jolla, California 92093

Ana Širović, Cochair
Texas A&M University Galveston, PO Box 1675, Galveston, Texas 77553

Invited Paper

8:00

2aAB1. Marine mammal distribution in the Gulf of Alaska from long-term passive acoustic monitoring. Ally Rice (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., #0205, La Jolla, CA 92093, ally.c.rice@gmail.com), Ana Širović (Texas A&M Univ. at Galveston, Galveston, TX), Jennifer Trickey, John Hildebrand, and Simone Baumann-Pickering (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

The Gulf of Alaska is an important habitat for a diverse array of marine mammals. Conservation of these populations requires understanding how they utilize different habitats. Passive acoustic monitoring can be used to study marine mammal distributions because the unique call types produced by many species are well documented. Between 2011 and 2015, a total of 3793 days of recordings were collected from autonomous acoustic recorders at five locations broadly distributed around the continental shelf, slope, and offshore seamounts in the Gulf of Alaska. The spatial and temporal patterns of seven cetacean species were examined and differences in habitat use and behavior were compared. Inshore habitats featured higher detections of humpback (*Megaptera novaeangliae*) and gray (*Eschrichtius robustus*) whales, while blue (*Balaenoptera musculus*) and sperm (*Physeter macrocephalus*) whales were more commonly detected offshore. The presence of most species was seasonal, with peak calling in spring and summer for gray and sperm whales, and fall and winter for fin (*B. physalus*) and humpback whales. Call type distributions within a species differed spatially for fin and killer (*Orcinus orca*) whales, and seasonally for blue whales. Spatial differences may indicate prey preferences, while seasonal changes are likely related to behavioral shifts from feeding to mating.

Contributed Papers

8:20

2aAB2. Using directional acoustics for mapping the spatial distribution of low frequency mammals. Bruce Martin (JASCO Appl. Sci., 32 Troop Ave., Ste. 202, Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com), Briand Gaudet (JASCO Appl. Sci., Dartmouth, NS, Canada), Corey Morris (NAFC, Dept. of Fisheries and Oceans, St. John's, NF, Canada), and Jennifer L. Miksis-Olds (CARE, Univ. of New Hampshire, Durham, NC)

Long-term passive acoustic monitoring programs have been conducted over several years using directional acoustic recorders. The data from these recorders provide an indication of the direction to each detected signal, which includes calls from blue, fin, sei, and minke whales. The directional results provide new information on the spatial distribution of these animals, including the type of habitat by each species as well as the number of animals vocalizing in an area. Examples will be presented using data recorded at the edge of the Grand Banks as well as from a site off Savanah Ga in waters 800 m deep.

8:35

2aAB3. Timing is everything: Drivers of interannual variability in blue whale migration. Angela R. Szesciorka (Marine Physical Lab., Scripps Inst. of Oceanogr., UC San Diego, 6944 Country Club Dr., La Jolla, CA 92037, angela@szesciorka.com), Lisa T. Ballance (NOAA Fisheries Service, Southwest Fisheries Sci. Ctr., La Jolla, CA), Ally Rice, John Hildebrand (Marine Physical Lab., Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA), Ana Širović (Texas A&M Univ. at Galveston, Galveston, TX), and Peter J. Franks (Integrative Oceanogr. Div., Scripps Inst. of Oceanogr., La Jolla, CA)

The growth and survival of large-bodied marine predators depend on temporal synchrony with resource availability. Baleen whales migrating long distances must therefore respond to interannual variability to avoid a predator-prey mismatch. Highly migratory and acoustically active, blue whales are a model species for investigating the drivers and timing of migration. Using passive acoustic recordings collected from 2007 to 2017, we examined the relationship among migration timing (inferred from blue

whale “D” and “B” calls), environmental indices, and measured krill abundance. Arrival to Southern California feeding grounds was correlated with sea surface temperatures (SST) on Costa Rica breeding grounds ($r=0.81$, <0.01). Colder SSTs in both regions resulted in early arrival and correlated with greater krill abundance in Southern California ($r^2=0.47$, $p=0.03$). The correlation between krill abundance and the transition from D to B calls ($r^2=0.55$, <0.01) suggested that in high-krill-abundance years, whales switched earlier from social to reproductive-related behavior on their feeding grounds. This phenotypic plasticity may allow blue whales to accommodate environmental variability while balancing important biological needs. However, D call onset dates increased significantly over the 11-year study ($r=0.68$, $p=0.03$), shifting arrival two months earlier. Longer time on the feeding grounds may increase ship-strike and entanglement risk for this endangered species.

8:50

2aAB4. Investigating fine scale frequency characteristics of Northeast Pacific blue whale B calls. Alexander W. Carbaugh-Rutland (Marine Biology, Texas A&M Galveston, 3428 Cove View Blvd Apartment 824, Galveston, TX 77554, alexc.rutland@tamu.edu), Arina Favilla (Ecology and Evolutionary Biology, Univ. of California, Santa Cruz, Santa Cruz, CA), and Ana Širović (Marine Biology, Texas A&M Galveston, Galveston, TX)

Passive acoustic monitoring is an effective tool for delineating population structure of blue whales *(Balaenoptera musculus).* Globally, there are at least nine regionally distinct blue whale songs, with at least two distinct groups within the North Pacific Ocean: the Northeast Pacific (NEP) and central or western Pacific populations. Investigation of the fine-scale frequency characteristics of the NEP blue whale song B unit was conducted from passive acoustic data collected between 2010 and 2013. Data were collected at two low latitude, putative breeding sites at Palmyra Atoll and the Hawaiian Islands and three higher latitude, feeding locations: off southern California, off Washington state, and in the Gulf of Alaska. Frequency measurements were extracted along the entire contour of B calls using a custom feature extraction tool in MATLAB. Data from these two different geographic and life-stage regions were compared to investigate possible fine-scale song separation within the larger region. At least two different variants of B unit were found and their geographic and temporal occurrence will be discussed.

9:05

2aAB5. Characterization of a bimodal call rate from tracked minke whales. Cameron R. Martin (Naval Information Warfare Ctr. Pacific, 53560 Hull St., San Diego, CA 92152, martincr@zoho.com), Stephen W. Martin (National Marine Mammal Foundation, San Diego, CA), Elizabeth Henderson, Tyler A. Helble (Naval Information Warfare Ctr. Pacific, San Diego, CA), Gabriela Alongi, and Brian Matsuyama (National Marine Mammal Foundation, San Diego, CA)

The minke whale call rate was characterized using 16 years of multichannel passive acoustic data collected from bottom-mounted hydrophones off the coast of Kauai, HI at the U.S. Navy Pacific Missile Range Facility’s instrumented range. Recorded data were post-processed for minke whale (*Balaenoptera acutorostrata*) boing vocalizations that were seasonally present using automated detection, classification, and localization algorithms. Semi-automated processes were used to perform spatio-temporal association of localizations to create individual whale tracks. The inter-call intervals (ICIs) from the resulting minke whale tracks exhibited a bimodal distribution. Preliminary results show that the longer ICI mean was 349.7 s, and the shorter or “rapid” ICI mean was 28.6 s. It is believed that only sexually mature males produce the boing call for breeding purposes, and the rapid ICI is hypothesized to be a challenge behavior exhibited when animals are within close proximity to each other. Data from February 2017 were examined in detail and contained four minke whale tracks with rapid ICIs, which appeared to occur when the nearest conspecific was within 10 km. Additional datasets were analyzed to further investigate distance between conspecifics relative to call rate.

9:20

2aAB6. Real-time passive acoustic monitoring results of the 2018 Pacific Region International Survey of Marine Megafauna off Western Canada. Thomas F. Norris (Bio-Waves, Inc., 517 Cornish Dr., Encinitas, CA 92024, thomas.f.norris@bio-waves.net), Nicholas Riddoch (Bio-Waves, Inc., Aberdeen, United Kingdom), Elizabeth T. Küsel (Bio-Waves, Inc., Portland, OR), Thomas Doniol-Valcroze, Robin Abernethy, and Linda Nichol (Fisheries and Oceans Canada, Nanaimo, Br. Columbia, Canada)

Real-time passive acoustic monitoring (PAM) was conducted as part of the first large-scale systematic survey of marine mammal distribution and abundance in territorial outer-coastal waters off western Canada. The survey was conducted for approximately five weeks, from early July to mid-August, 2018 using a 69-m Canadian Coast Guard and Fisheries Vessel. A four-element, towed hydrophone array and sonobuoys (type AN/SSQ-53F and AN/SSQ-53G) were deployed and monitored in real-time 24/7 by experienced bio-acousticians. The towed array was deployed for approximately 640 h (78% of the time underway). Approximately 145 independent detections of marine mammals were made. Sperm whales ($n=53$) and porpoises ($n= 49$) represented just over two-thirds of all detections. Ten near real-time localizations of sperm whales and 31 localization of porpoises were made, representing just over 60% of all ($n=67$) all high-quality localizations. Sonobuoy effort consisted of semi-systematic deployments day and night and when sightings of large baleen whales occurred. A total of 112 sonobuoys were deployed successfully and monitored in directional (DIFAR) mode. Marine mammal calls were recorded on 32% (36) of sonobuoys. Sonobuoy detections included killer whales calls and clicks, sperm whale clicks, dolphin clicks, fin whale calls, and blue whale calls (both B and D type calls).

9:35

2aAB7. Spatiotemporal distribution and population structure of sperm whales in the Eastern North Pacific. Natalie Posdaljian (Scripps Inst. of Oceanogr., 4581 39th St., Unit 7, San Diego, CA 92116, nposdalj@ucsd.edu), Alba Solsona Berga, Simone Baumann-Pickering, Kaitlin E. Frasier, and John Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA)

Sperm whales are a sexually dimorphic, cosmopolitan species, with a stratified distribution. Generally, males grow larger and travel to higher latitudes, while mature females and immature whales form social units thought to be more common in tropical and temperate regions. Little is known about their migration, especially by males for mating. Here, we outline an approach to detect presence of sperm whales echolocation clicks and the spatiotemporal distribution of sexes at 12 sites along the eastern North Pacific by using click characteristics as a proxy to estimate acoustic total body length and as an indicator for sex. Sperm whales were detected at all sites, including continental slope, deep abyssal, and seamount habitats. Some recording sites showed seasonal patterns potentially linked to the demographic composition of the population, especially males moving between temperate and high latitude regions. We also found females frequenting sites in the high-latitude habitats of the Gulf of Alaska and the Bering Sea Aleutian Islands, where they have been rarely documented since whaling. Our analysis improves the understanding of the spatiotemporal distribution and population structure of sperm whales which is essential for efficiently estimating densities to inform appropriate management and conservation measures.

9:50–10:05 Break

10:05

2aAB8. Call discrimination for an unknown pod of killer whales (*Orcinus orca*) in the Eastern Canadian Arctic. Jessica J. Sportelli (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 8635 Discovery way, Ritter Hall 200G, La Jolla, CA 92093, jsportel@ucsd.edu), Joshua M. Jones, Kaitlin E. Frasier, Ann Bowles (Hubbs Seaworld Res. Inst., San Diego, CA), John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), and Kristin H. Westdal (Oceans North, Winnipeg, MB, Canada)

Killer whales produce pulsed calls, which are used for communication. Calls are highly stereotyped, and repertoires are unique to individual pods. Discrimination amongst these calls and comparison of call repertoires

between pods can help determine population structure in killer whales and can be used to track pod movements. Calls were detected in underwater acoustic recordings from August to September 2017 in the waters near the community of Pond Inlet, Nunavut, Canada. Eight stereotypic call types were identified using whistle contour extraction and network analysis to compare contours. We present a repertoire of killer whale calls recorded. The potential for increased killer whale presence and magnitude of predation on narwhals is a source of concern for management of the population and by Inuit subsistence hunters who rely on narwhals for food and economic benefit. Describing the acoustic repertoire of killer whales seasonally present in the Canadian Arctic may help identify the stock or pod and determine their seasonal movements. Comparisons of this repertoire with killer whale calls from other Atlantic pods has not yet yielded a match. However, the results presented may provide a basis for future comparisons and aid in identifying killer whale ecotypes making seasonal incursions into Arctic waters.

10:20

2aAB9. Cetacean occurrence near the South Shetland Islands based on long-term passive acoustic monitoring. Jennifer S. Trickey (UCSD Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093, jtrickey@ucsd.edu), Ally Rice, Alba Solsona Berga, Ann Bartlett, Joshua M. Jones, Bruce Thayre, John Hurwitz, Simone Baumann-Pickering (UCSD Scripps Inst. of Oceanogr., La Jolla, CA), Ana Širović (Texas A&M Univ., Galveston, TX), John Hildebrand (UCSD Scripps Inst. of Oceanogr., La Jolla, CA), M. Vanesa Reyes Reyes, Marta Hevia, Alexander Marino, Mariana L. Melcon, and Miguel Iniguez (Fundacion Cethus, Olivos, Argentina)

Passive acoustic monitoring is an effective tool to investigate cetacean ecology, particularly in remote locations and over long time periods. Several cetacean species occur in Antarctic waters, but knowledge of their year-round relative abundance, distribution, and seasonality is relatively sparse. Long-term, broadband recordings were collected near the South Shetland Islands between 2014 and 2016, and spatio-temporal patterns in cetacean acoustic occurrence were investigated at three different locations within this area. Baleen whales detected included blue (*Balaenoptera musculus*), fin (*B. physalus*), and humpback whales (*Megaptera novaeangliae*), and several toothed whales, including sperm (*Physeter macrocephalus*) and killer whales (*Orcinus orca*) were also present in the datasets. Overall detection rates of these species varied across the recording sites and also showed strong seasonal trends. Sea ice dynamics will be explored as a possible driver for the shifts in seasonal relative abundance of cetaceans in this region of the Southern Ocean, and proximity to the Southern Antarctic Circumpolar Current Front (SACCF) will be examined to evaluate differences in habitat use across the sites.

10:35

2aAB10. Analysis and automatic detection of potential Omura's whale signals in the Indian Ocean. Scott D. Frank (Mathematics, Marist College, 3399 North Ave., Poughkeepsie, NY 12601, scott.frank@marist.edu), Michelle A. Bartolo (Mathematics, Marist College, Poughkeepsie, NY), and Danielle Harris (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St Andrews, United Kingdom)

The north Indian Ocean is a rich environment for passive acoustic monitoring of baleen whales including fin and blue whales, with several different populations of blue whales having been identified by recordings of their song type. A signal known as the Diego Garcia Croak (DGC) has been recorded on two hydrophone triads near Diego Garcia that make up part of the International Monitoring System of the Comprehensive Nuclear Test Ban Treaty Organization. The DGC signal was tentatively attributed to a blue whale (Sousa and Harris, 2015) but has recently been linked to the Omura's whale (Cerchio *et al.*, 2019). In previous work, the DGC has been observed visually on spectrograms and initial time and frequency statistics obtained. In this work, a least squares detector is constructed for the signals and applied to long term recordings from 2002 and 2003. The resulting detections are used to extend previous measurements of the DGC signal, generating a robust characterization of the DGC, including frequency statistics, inter-call intervals, and call lengths. Further, localization methods will be applied to initial time-difference of arrival results and used to explore source level estimation of the DGC signal.

10:50

2aAB11. Bowhead whale acoustic detection probability and spatial density of calls in ice-free and ice-covered Arctic waters. Joshua M. Jones (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA, j8jones@ucsd.edu), John Hildebrand, Bruce Thayre, and Sean M. Wiggins (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

Estimation of Arctic marine mammal presence and acoustic density from passive acoustic monitoring is complicated by effects of the environment and noise levels on sound propagation and the detection of calls. We used acoustic modeling and detection simulations to estimate site-specific transmission loss for sounds produced by bowhead whales in the northeast Chukchi Sea at two sites, outer shelf and continental slope, with and without sea ice cover at various ocean noise levels. A time series of hourly acoustic detection probability was produced for the two recording sites between 2012 and 2013 from modeled transmission loss for daily sea ice state and hourly average noise levels. We applied the detection probability to recorded bowhead whale call detections to correct for effects of ice cover and noise level. Corrected acoustic presence suggests a decrease in detectable calls within a 40 km radius of the recording sites with arrival of open water while the uncorrected detections increase or change little. When sea ice state is constant, substantial variability in acoustic presence tracks closely with changing noise levels. These results highlight the importance of incorporating effects of the environment and changing noise levels when interpreting results of passive acoustic monitoring.

11:05

2aAB12. Behavioral and environmental context of Antarctic minke whale vocalizations. Sarah Weindorf (UC Santa Cruz, P.O. Box 1227, Menlo Park, CA 94026, sgweindorf@gmail.com), Dave Cade (Stanford, Pacific Grove, CA), Caroline B. Casey, Ari Friedlaender (UC Santa Cruz, Santa Cruz, CA), Jeremy Goldbogen (Stanford, Pacific Grove, CA), Emma Levy, Jacob Linsky (UC Santa Cruz, Santa Cruz, CA), and Douglas Nowacek (Duke, Beaufort, NC)

Acoustic signaling is the predominant form of communication between cetaceans, and understanding the behavioral state of calling individuals can provide insights into their function. We used multi-sensor tags equipped with concurrent video and acoustic recorders to characterize the vocal repertoire and associated behavior of Antarctic minke whales around the Antarctic Peninsula. Whale vocalizations were sorted into distinct call types based on several known acoustic parameters including 90% call duration, frequency bandwidth, and frequency quartiles. Call types were then compared to concurrent video and sensor data to determine the behavioral state and environmental conditions surrounding vocalizing individuals. We found several distinct sound types ranging in frequency from 180–560 Hz, consisting of pulse trains, down-sweeps, growls, and other sound types not previously described. These sounds are produced during distinct behavioral states including foraging, traveling, socializing, and resting and are emitted both in solitary and in groups of up to five animals. Additionally, vocalizations were emitted in both open water and sea ice conditions. By combining acoustic recordings with animal-borne video, our results represent a significant advancement in our understanding of Antarctic minke whale behavior and improve our capacity to acoustically monitor this species in a rapidly changing Antarctic region.

11:20

2aAB13. Reliable lower bound for number of calling marine mammals in Chukchi Sea. John Spiesberger (Earth and Environ. Sci., Univ. of Pennsylvania, 240 S. 33rd St., Philadelphia, PA 19104-6316, johnsr@sas.upenn.edu), Catherine L. Berchok (Marine Mammal Lab., NOAA, Alaska Fisheries Sci. Ctr., Seattle, WA), Daniel Woodrich (Joint Inst. for the Study of the Atmosphere and Ocean, Univ. of Washington, Seattle, WA), Pranav G. Iyer (Dept of Phys. and Astronomy, Univ. of Pennsylvania, Philadelphia, PA), Alexander Schoeny (Dept. of Mathematics, Univ. of Pennsylvania, Philadelphia, PA), and Krishna Sivakumar (The Wharton School, Univ. of Pennsylvania, Philadelphia, PA)

A reliable lower bound is derived for the number of vocalizing bowhead whales and walruses in the Chukchi Sea from ten acoustic receivers deployed during NOAA's CHAOZ experiment in 2011. The lower bound

is derived from extremely reliable 100% confidence intervals (CI) of the sound's locations. CI are derived from measurements of the Time Differences of Arrival (TDOA) of sound between receivers. Sequential Bound Estimation (SBE) is used to estimate the CI by assimilating the TDOA. SBE was successfully and independently evaluated by the U.S. Navy. It is a nonlinear non-Bayesian technique capable of explicitly accounting for all errors affecting uncertainty while being computationally efficient. To date, its CI always contain the true location of the source. The CI are affected by uncertainty of sound speed, TDOA, receiver location, and clock time. For two-dimensional models of location, i.e., longitude/latitude, it appears necessary to utilize isodiachrons instead of hyperbolas to obtain reliable CI because of the extreme non-homogeneity of sound speed needed to obtain correct location. [Supported by the North Pacific Research Board and Microsoft.]

TUESDAY MORNING, 3 DECEMBER 2019 EMPRESS, 8:30 A.M. TO 9:35 A.M.

Session 2aAOa

Acoustical Oceanography and Underwater Acoustics: Session in Honor of Michael Buckingham I

Grant B. Deane, Cochair
Marine Physical Lab., Univ. of California, San Diego, La Jolla, California 92093

Simon E. Freeman, Cochair
Naval Undersea Warfare Ctr., Alexandria, Virginia 22310

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

Chair's Introduction—8:30

Invited Papers

8:35

2aAOa1. Mike Buckinghams's contributions to sub-Arctic acoustics. Orest Diachok (Poseidon Sound, 11100 Johns Hopkins Rd., Laurel, MD 20723, orestdia@aol.com)

Mike initiated his research in ocean acoustics and his love affair with aircraft during his tenure with the Royal Aerospace Establishment (RAE). Between 1984 and 1990, he led the RAE effort to characterize and model the ambient noise field in the sub-Arctic (Buckingham, 1987, 1990, and 1994). Mike deployed and monitored sonobuoys from aircraft at several distances from the ice edge. His experiment confirmed Diachok and Winokur's (1974) observations of a peak in ambient noise levels at the ice edge due to the interaction of ocean waves with sea ice. More importantly, Mike discovered an "enigmatic" intermittent source of high ambient noise levels, which peaked at about 20 Hz, at or near the Greenland glacier. He concluded that the source mechanism was associated with the glacier. In 1990, the Scripps Institution of Oceanography offered Mike a position, in part because of his stimulating pre-offer lecture on ambient noise in the sub-Arctic. As a result of increased melting of glaciers, noise generation at the glacier ocean boundary has become a "hot" research topic. The recent well-designed interdisciplinary experiment of Glowacki *et al.* (2015) revealed that calving and crashing of glacier blocks into the sea are the most likely cause of Mike's enigmatic noise.

8:55

2aAOa2. Ice floes' clashing as the source for acoustic ambient noise in the marginal ice zone of the Arctic Ocean. Chi-Fang Chen (Eng. Sci. and Ocean Eng., National Taiwan Univ., No. 1 Roosevelt Rd. Sec.#4, Taipei 106, Taiwan, chifang@ntu.edu.tw)

The ambient noise in the marginal ice zone (MIZ) of the Arctic Ocean is generated by the clashing of ice floes on the sea surface, which is a feasible source mechanism based on the experimental evidence. A theoretical model of the MIZ ambient noise based on a floe/floe collision mechanism is presented. It predicts time series and spectra whose features are broadly consistent with those observed in the data. The noise spectral density in the frequency band from 50 Hz to 1 kHz varies with frequency as f- n, where the index n shows an average value approximately equal to 2. Over a period of five days or so, n was observed to vary between a low value of 1.0 and a high value of 3.0. This work was done with Professor Michael Buckingham when he was a visiting professor in MIT in 1986.

9:15

2aAOa3. Benchmarking ocean acoustic models. Michael B. Porter (Heat, Light, and Sound Res., Inc., 12625 High Bluff Dr. Ste. 211, San Diego, CA 92130, Porter@hlsresearch.com)

In this special session honoring the work by Buckingham, we review some of his early research from the 1980s where he developed analytic solutions to various benchmark problems. In particular, he developed elegant solutions for (1) the perfect wedge; (2) a penetrable wedge; and (3) a conical seamount. These were very valuable at the time in resolving errors in early parabolic equation models; they have continued to be important as computational models for fully three-dimensional environments have become more widely used.

TUESDAY MORNING, 3 DECEMBER 2019 EMPRESS, 11:00 A.M. TO 12:00 NOON

Session 2aAOb

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

Peter Gerstoft, Chair

SIO Marine Phys Lab MC0238, Univ of California San Diego, 9500 Gillman Drive, La Jolla, California 92093

Chair's Introduction—11:00

Invited Paper

11:05

2aAOb1. The development of acoustic mapping of ocean currents in coastal seas. Chen-Fen Huang (Inst. of Oceanogr., National Taiwan Univ., No. 1, Sec. 4, Roosevelt Rd., Taipei 10617, Taiwan, chenfen@ntu.edu.tw)

Coastal oceanography is characterized by a high spatial and temporal variability, limiting the observational and predictive capability for environmental protection. Among various measurement techniques for studying ocean dynamics, the method of ocean acoustic tomography (OAT) is particularly useful for obtaining the spatial variation of current fields in the ocean interior. Traditionally, with only a limited number of the moored acoustic tomographic sensors, the inversion of currents in coastal areas still suffers insufficient sampling of the water volume. Recently, incorporating a ship-towed sensor has attracted considerable attention due to their capability of resolving the small spatial scale of ocean features. In this talk, we will review the fundamentals of moving ship tomography and signal/data processing algorithms for current estimation using the difference of reciprocal travel times. The central idea is to increase the number of ray paths that sample the ocean column at various angles, per the projection slice theorem. The concept has been further extended by using an autonomous underwater vehicle (AUV) or an unmanned surface vehicle (USV) as an alternative carrier for the sensor. This configuration has been demonstrated by recent experiments. This talk concludes with a discussion of future research activities and applications.

Session 2aBAa

Biomedical Acoustics: Application of Quantitative Ultrasound *In Vivo* in Humans I

Jonathan Mamou, Cochair
F. L. Lizzi Center for Biomedical Engineering, Riverside Research, 156 William St., 9th Floor, New York, New York 10038

Michael Oelze, Cochair
UIUC, 405 N Mathews, Urbana, Illinois 61801

Invited Papers

8:00

2aBAa1. High-frequency quantitative ultrasound and B-mode analysis for characterization of peripheral nerves including carpal tunnel syndrome. Michal Byra (Dept. of Ultrasound, Inst. of Fundamental Technol. Res., Polish Acad. of Sci., Warsaw, Poland), Jonathan Wong (Res. and Radiology Services, VA San Diego Healthcare System, San Diego, CA), Sameer Shah (Orthopedic Surgery and Bioeng., Univ. of California, San Diego, San Diego, CA), Aiguo Han, W. D. O'Brien, Jr. (Bioacoust. Res. Lab., Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), Jiang Du, Eric Chang (Radiology, Univ. of California, San Diego, San Diego, CA), and Michael P. Andre (Dept. of Radiology, Univ. of California, San Diego, and the San Diego VA Healthcare System, 9500 Gilman Dr., La Jolla, CA 92093, mandre@ucsd.edu)

We investigated the use of high-frequency quantitative ultrasound (QUS) and B-mode texture features to characterize ulnar and median nerve fascicles using a clinical scanner (Vevo MD) and a 30-MHz center-frequency probe. US correlation with histology was first investigated in the ulnar nerve *in situ* in cadaveric specimens. 85 fascicles were matched in B-mode images and the histology sections. Collagen and myelin concentrations were quantified from trichrome labeling, and backscatter coefficient (-24.89 $\pm$ 8.31 dB), attenuation coefficient (0.92 $\pm$ 0.04 dB/cm MHz), Nakagami parameter (1.01 $\pm$ 0.18) and entropy (6.92 $\pm$ 0.83) were calculated from ultrasound data. B-mode texture features were obtained via the gray-level co-occurrence matrix algorithm. Combined collagen and myelin concentration were significantly correlated with the backscatter coefficient (R = -0.68), entropy (R = -0.51), and several texture features. For the median nerve, we measured backscatter and morphology in 10 patients with carpal tunnel syndrome and 21 healthy volunteers. Significant differences (<0.01) between patients and controls and AUC 0.89–0.94 for QUS biomarkers were observed. Our study indicates that QUS may potentially provide useful information on structural components of even very small nerves (2 $\times$ 4 mm) and fascicles for diagnosing and monitoring injury, and surgical planning.

8:20

2aBAa2. Assessment of hepatic steatosis in humans using multivariable quantitative ultrasound. Aiguo Han (Univ. of Illinois, Urbana, IL), Yingzhen N. Zhang, Andrew S. Boehringer, Vivian Montes, Michael P. Andre (Univ. of California at San Diego, La Jolla, CA), John W. Erdman (Univ. of Illinois, Urbana, IL), Rohit Loomba, Mark A. Valasek, Claude B. Sirlin (Univ. of California at San Diego, La Jolla, CA), and W. D. O'Brien, Jr. (Univ. of Illinois, 405 N. Mathews, Urbana, IL 61801, wdo@uiuc.edu)

The objective of soft-tissue quantitative ultrasound (QUS) is to improve diagnostic ultrasound imaging via quantitative outcomes. Over the past three or so decades, there have been an increasing number of QUS successes. In a UIUC-UCSD collaboration, nonalcoholic fatty liver disease (NAFLD) assessed from seven QUS biomarkers [AC, BSC, three Lizzi-Feleppa markers (slope, intercept, midband), two envelope parameters (k and mu)] derived from ultrasound RF data shows dependencies with the liver fat content in human subjects. 102 participants underwent QUS exams on the right liver lobe with an Acuson S3000 scanner (4C1 and 6C1HD transducers). Two multivariable models have been developed based on QUS biomarkers: (1) generalized linear regression model to predict hepatic PDFF using stepwise regression for biomarker selection and (2) regularized logistic regression model to classify normal (MRI-PDFF <5%, n = 26) versus NAFLD (MRI-PDFF $\geq$ 5%) using LASSO regularization for biomarker selection. Leave-one-out cross-validation was used for both models. The regression model selected the midband and k-parameter (R^2 = 0.59, Spearman ρ = 0.84, and Pearson's r = 0.77). The classifier model selected the midband and μ-parameter (AUC of 0.89). Multivariable QUS provides higher quantification and classification accuracy than univariate QUS approaches. [R01DK106419 and Siemens Healthineers.]

8:40

2aBAa3. Deep learning approaches for quantitative analysis of ultrasound backscattered signals. Aiguo Han (Bioacoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 306 North Wright St., Urbana, IL 61801, han51@uiuc.edu), Andrew S. Boehringer, Vivian Montes (Liver Imaging Group, Dept. of Radiology, Univ. of California, San Diego, La Jolla, CA), Michael P. Andre (Dept. of Radiology, Univ. of California, San Diego, La Jolla, CA), John W. Erdman (Dept. of Food Sci. and Human Nutrition, Univ. of Illinois at Urbana-Champaign, Urbana, IL), Rohit Loomba (NAFLD Res. Ctr., Div. of Gastroenterology, Dept. of Medicine, Univ. of California, San Diego, La Jolla, CA), Mark A. Valasek (Dept. of Pathol., Univ. of California, San Diego, CA), Claude B. Sirlin (Liver Imaging Group, Dept. of Radiology, Univ. of California, San Diego, La Jolla, CA), and W. D. O'Brien, Jr. (Bioacoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Quantitative ultrasound (QUS) aims to improve diagnostic ultrasound imaging by extracting objective tissue parameters from backscattered signals. Deep learning can facilitate this process because of its ability to extract high-level information from the raw data. We have demonstrated that deep learning approaches applied to backscattered signals can accurately quantify liver fat noninvasively. We will discuss three deep learning approaches herein: (1) a one-dimensional convolutional neural network (1-D-CNN) for uncalibrated radiofrequency (RF) data; (2) a two-dimensional CNN (2-D-CNN) for uncalibrated spectrograms; and (3) a 2-D-CNN for calibrated spectrograms. Approaches (1) and (2) do not require system calibration with a physical phantom, whereas Approach (3) obtains calibrated spectrograms with a physical phantom. Each approach is evaluated using three datasets that contain ultrasound RF data acquired from the right liver lobe and contemporaneous MRI-PDFF (reference standard) in participants with and without nonalcoholic fatty liver disease: 204 scanned by a Siemens S2000 ultrasound scanner (4C1), 70 scanned by a Siemens S3000 scanner (4C1 and/or 6C1HD), and 104 scanned by both the S3000 and a GE Logiq e9 scanner (C1-6). We will discuss the advantages and disadvantages of each approach evaluated using data acquired across multiple ultrasound scanner platforms. [Supported by R01DK106419.]

9:00

2aBAa4. Evaluation of reference-free and model-free spectral-based quantitative ultrasound *in vivo*. Michael L. Oelze (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 405 N Mathews, Urbana, IL 61801, oelze@uiuc.edu) and Trong N. Nguyen (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Two limitations of spectral-based quantitative ultrasound techniques for characterizing tissues *in vivo* are: (1) the need for one or more separate calibration measurements per scanning session and (2) lack of well-defined models of scattering. In the first case, the requirement to take a reference scan for each setting used in a clinical procedure can disrupt the busy clinical workflow. In the second case, the choice of models to interpret backscatter coefficient data may not utilize all of the information available in the backscatter coefficient spectrum. To address these issues, we have explored the use of machine learning techniques in phantoms and *in vivo* without requiring a model and in some cases without requiring a reference. A convolutional neural network (CNN) was utilized that took in the raw RF ultrasonic backscattered data from the livers of rabbits (N = 57) maintained on a high fat diet and classifier was constructed from the CNN. The same scanner settings were used on all rabbits and no reference signal was used with the CNN. The hypothesis was that the CNN could differentiate between tissue signal and system signal. The accuracies of the reference-free and model-free approaches were compared to traditional quantitative ultrasound techniques.

9:20

2aBAa5. Tissue characterization using backscatter and attenuation coefficients: theoretical developments and *in vivo* applications. Roberto J. Lavarello (Departamento de Ingeniería, Pontificia Universidad Catolica del Peru, Av. Universitaria 1801, San Miguel, Lima 32, Peru, lavarello.rj@pucp.edu.pe)

Conventional echographic imaging depicts anatomical structure by displaying the magnitude of backscattered ultrasound echoes. However, ultrasonic radiofrequency data contains a richer information content that can be exploited for constructing images of intrinsic tissue properties. In particular, spectral-based ultrasonic tissue characterization techniques allow imaging parameters such as the backscatter (BSC) and attenuation (AC) coefficients. Even though this type of analysis has been explored for decades, several challenges ranging from technical algorithmic issues to the lack of widely validated, successful clinical applications have limited efforts directed towards these imaging modalities. The formulation of BSC and AC estimation as a 2-D inverse problem in combination with regularization methods is discussed, and this approach is shown to significantly extend the trade-off between estimation precision and spatial resolution. Recent applications of BSC and AC estimation *in vivo* are also discussed, including the assessment of interstitial fibrosis and tubular atrophy (IFTA) and inflammation in renal allografts, the diagnosis of cervical lymph nodes, and the high frequency characterization of skin.

9:40

2aBAa6. Quantitative ultrasound in obstetrics and perinatal care. Ivan M. Rosado-Mendez (Instituto de Fisica, Universidad Nacional Autonoma de Mexico, Circuito de la Investigacion Cientifica s/n, Ciudad Universitaria, Coyoacan 03310, Mexico, irosado@fisica.unam.mx), Abel Torres, Laura Castañeda-Martinez, Francisco Torres-Arvizu (Instituto de Fisica, Universidad Nacional Autonoma de Mexico, Coyoacan, CDMX, Mexico), Lindsey Carlson (Medical Phys., Univ. of Wisconsin, Madison, WI), Mark L. Palmeri (Biomedical Eng., Duke Univ., Durham, NC), James Zagzebski (Medical Phys., Univ. of Wisconsin, Madison, WI), Hassan Rivaz (Elec. and Comput. Eng., Concordia Univ., Montreal, PQ, Canada), Chrysanthy Ikonomidou (Dept. of Neurology, Univ. of Wisconsin, Madison, WI), Helen Feltovich (Maternal Fetal Medicine, Intermountain Healthcare, Provo, UT), and Timothy J. Hall (Medical Phys., Univ. of Wisconsin, Madison, WI)

This talk will present recent results from our multi-institutional effort to advance the application of Quantitative Ultrasound in obstetrics and in perinatal care. In the field of obstetrics, we are investigating the use of shear wave elasticity imaging (SWEI) to predict the risk of preterm birth based on assessing the viscoelastic properties of the uterine cervix. Longitudinal studies in non-human primates (NHP) and humans showed a similar rate of decrease of the shear wave speed, SWS (1.6% and 1.5%/% gestational age, respectively), validating the animal model and indicating the value of SWS change as a biomarker for cervical softening. We are currently optimizing the analysis of shear wave dispersion to assess changes in viscosity. In the field of perinatal care, results from a pilot study in NHP

neonates suggest that the effective size of diffuse scatterers can be used as a biomarker for thalamic apoptosis induced by long exposures to anesthesia. We are currently investigating the use of coherent scattering features to increase the sensitivity and specificity to neuroapoptosis and to improve attenuation and backscatter coefficient estimations in complex tissues such as the cervix and the neonate brain. [Acknowledgments: All protocols have been IRB- and/or IACUC-approved, and are HIPAA compliant. Equipment loan and technical support from Siemens Healthcare. Funded by: NIH Grant Nos. T32CA009206, F31HD082911, R01HD072077, UL1TR000427, P51OD011106, and R01HD083001; NSERC Grant No. RGPIN-2015-04136; and UNAM-PAPIIT IA104518 and IN107916.

10:00–10:15 Break

10:15

2aBAa7. Quantitative multiparametric ultrasound for prostate cancer targeted biopsy. D. Cody Morris (Biomedical Eng., Duke Univ., 101 Sci. Dr., Durham, NC 27708, cody.morris@duke.edu), Derek Y. Chan (Biomedical Eng., Duke Univ., Durham, NC), Hong Chen (Frederec L. Lizzi Ctr. for Biomedical Eng., Riverside Res. Inst., New York, NY), Mark L. Palmeri (Biomedical Eng., Duke Univ., Durham, NC), Thomas Polascik, Rajan T. Gupta (Duke Cancer Inst., Durham, NC), Wen-Chi Foo, Jiaoti Huang (Pathol., Duke Health, Durham, NC), Jonathan Mamou (Frederec L. Lizzi Ctr. for Biomedical Eng., Riverside Res. Inst., New York, NY), and Kathryn Nightingale (Biomedical Eng., Duke Univ., Durham, NC)

Prostate cancer (PCa) is conventionally diagnosed using transrectal ultrasound (TRUS) guided biopsy. B-mode ultrasound is used to direct the biopsy needle to regions of the prostate gland, but is not used to target cancer-suspicious regions due to its low specificity for PCa. We are developing multiparametric quantitative ultrasound imaging approaches to facilitate targeting ultrasound-guided biopsies toward regions suspicious for PCa. To identify PCa during TRUS imaging, we combine information from four ultrasonic techniques: acoustic radiation force impulse (ARFI) imaging, shear wave elasticity imaging (SWEI), quantitative ultrasound (QUS) and B-mode ultrasound. 3-D co-registered *in vivo* ultrasound data, and MRI T2-weighted and ADC data volumes have been acquired from 30 patients prior to radical prostatectomy, obtaining estimates of ARFI displacement, shear wave speed, QUS midband fit, B-mode brightness, T2 intensity, and ADC values. In each data volume, healthy and cancerous regions were manually segmented with guidance from whole mount histology; the intersection of the cancerous segmentations from all modalities was labeled as the ground truth. We will present a range of classification approaches, including LDA and SVM, to combine information from each modality in order to develop automated tools for targeted biopsy.

10:35

2aBAa8. Quantitative ultrasound to better evaluate heterogeneous tumor response to therapy. Lori Bridal (Laboratoire Imagerie Biomedicale - CNRS, INSERM, SU, 15 rue de l'ecole de medecine, Paris 75006, France, lori.bridal@upmc.fr), Alain Coron, and Jérôme Gâteau (Laboratoire Imagerie Biomedicale - CNRS, INSERM, SU, Paris, France)

Heterogenity of tumor structure and function (e.g., stromal density, vascularization, and oxygen distribution) can influence local cellular proliferation and therapeutic resistance. Emerging ultrasound imaging techniques can map parameters related to these features but such approaches are not yet in standard clinical practice. The goal of our work is to develop robust protocols to provide more complete information during therapy by quantitatively mapping and interpreting tumor heterogeneity using emerging ultrasonic techniques. Within murine tumor models Shear Wave Elastography (SWE), Contrast-Enhanced Ultrasound (CEUS) and Quantitative Ultrasound (QUS) techniques have been applied with clinical imaging systems to compare changes observed during anti-angiogenic and cytotoxic therapy as compared to controls. In a Lewis Lung Carcinoma model, mice receiving an anti-angiogenic therapy had the highest levels of necrosis and fibrosis and presented the highest number of correlations between quantitative parameters related to microvascular function and tumor structure. This indirectly demonstrates, for this tumor-type, that quantitative parameters measured with clinical imaging systems can help differentiate tumor therapeutic responses. Promising results with ultrasound localization microscopy and photoacoustics will also be reviewed because of the important new information they offer. Finally, the path towards integration of these quantitative ultrasound techniques in clinical practice will be discussed.

10:55

2aBAa9. Quantitative evaluation of liver diseases using multi-PDF models. Tadashi Yamaguchi (Chiba Univ., 1-33 Yayoicho, Inage, Chiba 2638522, Japan, yamaguchi@faculty.chiba-u.jp)

We have proposed the triple Rayleigh probability density function (TR-PDF) model as the *in vivo* QUS method of liver fibrosis. Additionally, the double Nakagami probability density function (DN-PDF) model was proposed to evaluate the amount and distribution condition of fat droplets in the fatty-liver livers. The DN-PDF permitted to detect early and moderate fatty livers in previous *ex vivo* studies with over 15 MHz ultrasound. In our recent studies, we are examining clinical data to determine whether two multi-PDF models can evaluate each disease even at low frequencies. In this talk, the results of *in vivo* studies for liver fibrosis and fatty liver will be introduced.

11:15

2aBAa10. *In-vivo*-quantitative-ultrasound assessment of thyroid nodules. Jonathan Mamou (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, jmamou@riversideresearch.org), Poorani Goundan (Section of Endocrinology, Boston Univ. School of Medicine, Boston Medical Ctr., Boston, MA), Daniel Rohrbach (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York City, NY), Harshal Patel (Section of Endocrinology, Boston Univ. School of Medicine, Boston Medical Ctr., Boston, MA), Ernest Feleppa (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY), and Stephanie Lee (Section of Endocrinology, Boston Univ. School of Medicine, Boston Medical Ctr., Boston, MA)

Thyroid nodules suspicious for malignancy during clinical-ultrasound (US) evaluation undergo fine-needle-aspiration biopsy (FNAB) followed by cytological evaluation and, in indeterminate cases, an additional surgical intervention or molecular testing is recommended. Quantitative ultrasound (QUS) was investigated as a non-invasive approach to detect thyroid cancer and potentially reduce

the currently large number of unnecessary FNABs. RF data were acquired from 225 nodules of 208 patients using a GE Logiq E9 US equipped with a 10-MHz linear-array probe. Five QUS estimates were computed throughout each nodule. Specifically, effective scatter diameter, effective scatter concentration, midband fit, and intercept were obtained using a reference-phantom method. The Nakagami shape parameter was computed using a maximum-likelihood estimator. Mean and standard deviations (SD) of QUS estimates within nodules were used for classification. QUS-based classification performance was quantified using area-under-the-curve (AUC) values obtained from receiver-operating- characteristic analyses. An AUC value of 0.77 was obtained using the SD of intercept alone; the value increased to 0.86 when five QUS estimates were linearly combined. Our *in vivo* results demonstrate the potential of QUS methods to detect cancerous nodules. Ultimately, the clinical protocol could be altered to prevent FNABs of nodules highly unlikely to be malignant based on QUS analysis.

11:35

2aBAa11. Quantitative cardiac blood flow imaging with high frame rate ultrasound. Hideyuki Hasegawa (Univ. of Toyama, 3190 Gofuku, Toyama 9308555, Japan, hasegawa@eng.u-toyama.ac.jp) and Ryo Nagaoka (Univ. of Toyama, Toyama, Japan)

The measurement of blood flow is important for evaluation of cardiac function. Ultrasonic color Doppler imaging is one of the most frequently used methods for the measurement of cardiac blood flow. However, the Doppler method provides only the axial velocity components in the direction of ultrasound propagation. To overcome such a problem, the vector flow mapping method (VFM), which estimates 2-D blood flow velocity by applying the theory of fluid dynamics to the axial velocity field obtained by the color Doppler method. However, the frame rate is limited to less than 30 frames per second (fps), and better temporal resolution is preferable for evaluation of the rapid temporal change in cardiac blood flow. In the present study, echoes from blood cells were visualized by high frame rate ultrasound imaging at a frame rate of 6250 fps. Recently, echo particle image velocimetry (e-PIV) was developed for observation of cardiac blood flow. This method requires intravenous injection of ultrasonic contrast agents. On the other hand, the proposed method does not require contrast agents. Also, echoes from blood cells are visualized at a very high frame rate, and thus, blood flow velocity vectors can be estimated quantitatively by applying motion estimator to visualized echoes.

TUESDAY MORNING, 3 DECEMBER 2019 HANOVER, 9:00 A.M. TO 11:15 A.M.

Session 2aBAb

Biomedical Acoustics and Signal Processing in Acoustics: High Frame Rate Ultrasound Imaging: Technical Developments and Clinical Applications I

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

Alessandro Ramalli, Cochair
Cardiovascular Imaging and Dynamics, KU Leuven, Leuven 3000, Belgium

Chair's Introduction—9:00

Invited Papers

9:05

2aBAb1. Intrinsic cardiovascular strain and wave imaging using rapid frame rate imaging. Elisa Konofagou (Columbia Univ., 1210 Amsterdam Ave., ET351, New York, NY 10027, ek2191@columbia.edu)

Cardiovascular disease currently claims more lives each year in both men and women than the next two most deadly diseases combined, i.e., cancer and chronic lower respiratory disease. Among the cardiovascular diseases, coronary artery disease (CAD) is by far the most deadly accounting for 44% of cardiovascular-related deaths and causing approximately 1 of every 6 deaths in the United States in 2019. Approximately every 34 s, 1 American has a coronary event, and approximately every 1 min 23 s, an American will die of one. Clinical observations also show that the formation, expansion, and rupture phases of a vessel due to pathologies such as aneurysms and atherosclerotic plaques are each associated with changes in arterial stiffness. Pulse Wave Velocity (PWV) is often used as a biomarker, but imaging of pulse waves and characterizing the propagation has been essential in staging vascular disease. In this paper, ultrasound-based methodologies at high frame rates will be presented that provide information on both the mechanical and electrical properties of the myocardium as well as mechanical properties of the vascular wall through intrinsic phenomena and rapid wave propagation in order to better image the onset and progression of the aforementioned diseases.

9:25

2aBAb2. Ultrafast ultrasound breast scanning for improved image quality and functional information in 3-D. Gijs Hendriks, Chuan Chen, Hendrik Hansen (Radiology and Nuclear Medicine, Radboud Univ. Medical Ctr., Nijmegen, The Netherlands), and Chris L. de Korte (Radiology and Nuclear Medicine, Radboud Univ. Medical Ctr., MUSIC 766, PO Box 9101, Nijmegen 6500HB, The Netherlands, chris.dekorte@radboudumc.nl)

With an automated breast volume scanner (ABVS), a volumetric breast image can be constructed. Clinical studies show high sensitivity but also high recall-rates due to the detection of many lesions of uncertain malignant potential. It is known that breast lesions have different mechanical properties with respect to its surrounding tissue. Consequently, imaging the biomechanics of a breast might improve diagnosis in this disease. First, we improved the image quality by taking full advantage of the ultrafast acquisition and by applying dedicated signal processing techniques. Next, by imaging biomechanics we increased the specificity since compared to benign lesions, malignant lesions are often stiffer, and more grown into the surrounding tissue (firmly bonded). We extended conventional 2-D strain imaging to full 3-D (shear) strain imaging by combining automated breast volume scanning with ultrafast plane wave imaging (10 000 frames/s). Validation studies in breast phantoms revealed that lesions could be differentiated based on strain and shear strain imaging. Decreased strain values were found in stiff lesions and firmly bonded lesions demonstrated low shear strain values. Initial acquisitions in patients demonstrated decreasing strain values with increasing severity of the disease. Additionally, we developed an ultrasensitive 3-D Doppler method to detect neo-vascularization that might be present around malignant lesions. In conclusion, imaging the biomechanical properties in 3-D improves diagnosis breast cancer.

Contributed Papers

9:45

2aBAb3. Heterogeneous angular spectrum method for trans-skull imaging and focusing. Scott J. Schoen (Mech. Eng., Georgia Inst. of Technol., 901 Atlantic Dr. NW, Rm. 4125K, Atlanta, GA 30318, scottschoenjr@gatech.edu) and Costas Arvanitis (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Ultrasound has emerged as a promising modality for therapy and imaging of brain diseases. Existing methods for the correction of skull-related distortions (aberration) require significant computation, which hampers their use in real-time applications, including adaptive focusing and tracking of bubble dynamics when used with microbubble contrast agents. Unfortunately, the angular spectrum (AS) method, which is the most computationally efficient focusing/beamforming method, does not intrinsically account for heterogeneity in the propagation medium. Here, we present a full solution for the AS in a heterogeneous medium, and an analytical solution for the special case of a stratified medium. Simulations of trans-skull acoustic propagation were performed for passive acoustic mapping (PAM) and focal aberration correction. Results show the general solution provides accurate trans-skull focusing as compared to the homogeneous case (error 0.65 $\pm$ 0.27 mm) for clinically relevant frequencies (0.5 to 1.5 MHz). Source localization error was reduced by 70% (from 2.89 $\pm$ 1.76 mm to 0.68 $\pm$ 52 mm), and computation of the corrections required milliseconds (166 $\pm$ 37 ms, compared with 44 $\pm$ 4 ms; the analytical stratified solution is at least 54% faster than the full correction). The proposed phase correction method may provide a computationally efficient method for improved trans-skull focusing and imaging for real-time applications.

10:00–10:15 Break

10:15

2aBAb4. Real-time imaging of brain displacement during FUS neuromodulation in rodents *in vivo*. Tara Kugelman (Biomedical Eng., Columbia Univ., 630 W 168th St., New York, NY 10032, tk2694@columbia.edu), Mark T. Burgess, and Elisa Konofagou (Biomedical Eng., Columbia Univ., New York, NY)

Focused ultrasound (FUS) is capable of modulating the central and peripheral nervous system. Temperature increase, cavitation, and acoustic radiation force have been proposed as potential mechanisms. Thus, tissue displacement imaging during FUS modulation is critical to understanding its physical mechanism. A cranial window was formed to perform imaging during FUS neuromodulation in C57BL/6 mice. FUS sonications were performed with a 4 MHz single-element transducer with a focal volume 0.2 $\times$ 0.9 mm (Sonic Concepts, WA), an applied tone burst of 0.5 ms and peak positive pressures varying from 1.9 to 6.7 MPa. Real-time channel data using plane wave sequences were acquired with an 18 MHz linear array (Vermon, France) and axial displacements throughout the entire brain were estimated using 1D cross-correlation (9λ window, 99% overlap) at 1 kHz frame rate. Downward displacements were only detected during FUS within the focal region. During modulation, the cumulative displacement ranged from 0.021 $\pm$ 0.017 μm (n = 3) at 3.1 MPa to 0.24 $\pm$ 0.004 μm at 6.7 MPa (n = 3). Displacement decreased post-FUS and returned to null within 1–2 ms. Unlike in a preliminary phantom study, no shear waves were detected. Real-time displacement imaging could thus provide effective targeting and monitoring as well as unveil the mechanism during FUS modulation.

10:30

2aBAb5. Computational and ultrafast ultrasound assessment of arterial bi-directional stiffness from spontaneous pulsatile waves. Dan Ran (Dept. of Elec. and Electron. Eng., The Univ. of Hong Kong, Rm. 206, Chow Yei Ching Bldg., Pokfulam Rd., Hong Kong, Hong Kong 999077, China, randan@eee.hku.hk), JinPing Dong, He Li, and Wei-Ning Lee (Dept. of Elec. and Electron. Eng., The Univ. of Hong Kong, Hong Kong, Hong Kong)

Ultrafast vascular ultrasound imaging (UVUI) enables the capture of transient bi-directional arterial wall motions induced by pulsatile blood ejection. In this study, from fluid-structure-interaction (FSI) simulation and *in vitro* experiments, we quantitatively examined our recently observed longitudinally propagating natural wave, whose velocity (V_L) was tracked from the longitudinal arterial wall motion and thus assumed to reflect longitudinal stiffness. This supplements the well-established pulse wave, whose velocity (V_P) is estimated from radial motion and linked to transverse stiffness via the Moens-Korteweg equation. Three-dimensional two-way-coupled FSI simulation was created in COMSOL. The geometric parameters and loading condition of the simulated isotropic vessel model and a home-made vessel-mimicking phantom (radius: 6.43 mm; thickness: 2.07 mm; pressure: 10–25 mmHg) were the same. The phantom was connected to pulsatile flows; its ultrasound radiofrequency data were acquired by a Verasonics Vantage system at 6000 frames/s and then analyzed using the spatial-angular-compounding technique to obtain bi-directional displacements. V_P and V_L were estimated from radial and longitudinal wall displacements, respectively. Comparable V_L-to-V_P ratios in the simulation (2.83) and phantom (3.16 $\pm$ 0.66) cross-validated the inherent velocity relationship between the two waves in isotropic vessel wall, implying the potential of adopting UVUI to evaluate bi-directional arterial stiffness orientation from spontaneous pulsatile waves.

10:45

2aBAb6. Reconstruction of shear wave speed in layered media using time-harmonic excitation and pulse-echo imaging. E. G. Sunethra Dayavansha (Biomedical Eng., Univ. of Cincinnati, 3960 Cardiovascular Res. Ctr., 231 Albert Sabin Way, Cincinnati, OH 45267-0586, dayavaek@ucmail.uc.edu), Peter D. Grimm (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH), Jack Rubinstein (Medicine, Univ. of Cincinnati, Cincinnati, OH), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Recent advances in high-frequency ultrasound imaging have opened new possibilities for quantitative characterization of skin properties. Noninvasive, accurate measurements of elasticity for multiple layers (e.g., stratum corneum, epidermis, dermis) are potentially beneficial for development and validation of new skin treatments. Here, pulse-echo ultrasound imaging using linear arrays is used to measure time-harmonic tissue displacements induced by a mechanical shaker. It is shown that reconstruction of time-harmonic displacement amplitude and phase in induced elastic waves is feasible for frame rates exceeding the Nyquist frequency and for echoes acquired using multiple transmits per frame. Measured wavefields are inverted using spatial Fourier analysis, numerical solution of the Helmholtz equation by finite difference methods, and direct observation of shear wavelengths. These approaches are validated using a 5 MHz diagnostic imaging array with a clinical scanner (Zonare z.one, L8-1 array) and multlilayered phantoms with varying elastic properties and echogenicity. Further verification of these methods is performed by comparing with finite-element simulations employing a linear elasticity model. Feasibility for measurement of skin layer elastic properties using high-frequency ultrasonic imaging is demonstrated using a 50 MHz linear array and a small-animal scanner (Vevo 2100, MS700 array) for measurements on porcine skin.

11:00

2aBAb7. Blood viscosity could explain shear wave dispersion in the liver. Johannes Aichele (U1032, INSERM Labtau Univ. of Lyon, Cours Albert Thomas 152, Lyon 69003, France, johannes.aichele@inserm.fr) and Stefan Catheline (U1032, INSERM Labtau Univ. of Lyon, Lyon, France)

We recently showed that the observed shear wave dispersion in a soft, porous, water-saturated tissue can be explained by Biot's theory of poroelasticity. The theory explains the shear wave velocity increase with frequency due to a relative movement between the solid and the viscous fluid. We propose that fluid-solid interaction explains the observed shear wave dispersion in the liver, a naturally saturated organ. The liver is drawn through by a network of blood vessels and exposes a total porosity of about 14% [1]. Blood viscosity changes from patient to patient and depends on different factors such as hydration and fitness. We included the blood viscosity for a 14% porosity liver into the elasticity estimation from shear wave speed measurements for a given shear wave elastography dataset [2]. For 11 out of 50 patients, the fibrosis classification would change if blood viscosity is included. [1] C. Debbaut *et al.*, "Perfusion characteristics of the human hepatic microcirculation based on three-dimensional reconstructions and computational fluid dynamic analysis," *J. Biomech. Eng.* **134**(1), 011003 (2012). [2] Jang *et al.*, "Hemorheological alteration in patients clinically diagnosed with chronic liver diseases," *J. Korean Med. Sci.* **31**(12), 1943–1948 (2016).

TUESDAY MORNING, 3 DECEMBER 2019 — STUART, 7:45 A.M. TO 11:50 A.M.

Session 2aEA

Engineering Acoustics and Structural Acoustics and Vibration: Acoustical Engineering in Consumer Electronics

Edward Okorn, Cochair
GRAS NA, Inc., 2234 E. Enterprize Pkwy., Twinsburg, Ohio 44087

Caleb F. Sieck, Cochair
Code 7160, U.S. Naval Research Laboratory, 4555 Overlook Ave. SW, Washington, DC 20375

Chair's Introduction—7:45

Invited Papers

7:50

2aEA1. An interview with Jerry Harvey, inventor of the In-Ear-Monitor ~ born of Rock Stars!. Edward Okorn (GRAS NA, Inc., Twinsburg, OH) and Jerry Harvey (JH Audio, 111 West Jefferson St., #300, Orlando, FL 32801, Jerry@JHAudio.com)

While this is not a traditional academic presentation, it is an informal and open discussion with the inventor of the In-Ear-Monitor (IEM), by an untrained acoustician Jerry Harvey, out of the request to save the hearing of legendary Rock Star Alex Van Halen. In 1995, Harvey began tinkering and looking for components to solve the problem; in Japan, he found tiny electrical components; and in the United States, he found a tiny speaker designed for a pacemaker. He created a tiny in-ear speaker system that connected to a small receiver on Alex's belt via thin cables. The receiver then picked up the wireless signal from Harvey's mixing board. The in-ear monitors had two small speakers to separate output volume into low and high frequencies for bass and treble, and they fit into shells that were impressions of Alex's ears. The IEMs also blocked out ambient noise, and according to Alex Van Halen, "It was like night and day." Today IEMs are widely used by musicians, audio engineers, audiophiles, and television presenters in order to receive vocal instructions, from a producer that only the presenter hears. The IEM market is expected to reach close to $1 Billion by 2024.

8:20

2aEA2. Acoustic test fixtures: From KEMAR and beyond!. Edward Okorn (North America, GRAS Sound & Vib. - NA, Twinsburg, OH) and Peter Wulf-Andersen (GRAS Sound & Vib. A/S, Skovlytoften 33, Holte DK-2840, Denmark, PWA@GRAS.dk)

In 1972, KEMAR was introduced to the world. Based on the average of about 5000 males and females from the U.S. Air Force. It was the first head and torso simulator designed especially for acoustic research and enabled the hearing aid laboratories to perform simulated *in-situ* measurements of hearing aids. Today, more than 47 years after its introduction, KEMAR is able to test any device that contains both loudspeakers and microphones as well as performing binaural recordings of product sound and music. Fixturing, ear simulators, and special microphones, also known as couplers, are used by acousticians and engineers to design and test the many forms of modern acoustic electronics used from entertainment (Ear Buds, Headphones, IEMs, Smart Speakers, etc.), to hearing protection (Noise Cancelation), to medical devices (Hearing and Augmentation and Aids), to automotive cabins and sound systems, to military headsets alike. This presentation will discuss the evolution, ongoing design challenges, and future direction of Acoustic Test Fixtures required for the acoustic discovery and development going forward.

8:50

2aEA3. The perception and measurement of headphone sound quality. Sean E. Olive (Harman X, Harman Int., 8500 Balboa Blvd, Northridge, CA 91377, sean.olive@harman.com), Todd Welti, and Omid Khonsariopour (Harman X, Harman Int., Northridge, CA)

We recently completed a 7-year research project aimed at understanding the perception and measurement of headphone sound quality. A virtual headphone listening test method was developed to provide controlled, double-blind comparisons of different models of headphones and target response curves using a large number of trained and untrained listeners in USA, Canada, Germany, and China. From these data, we identified a new headphone target curve that is preferred by the majority of listeners. Statistical models were developed that predict listeners' headphone sound quality ratings based on objective headphone measurements. More recently, cluster analysis of headphone listening test data has shown there are three segments or classes of listeners based on similarities in their headphone sound preferences. Both demographic (i.e., age, gender, listening experience) and acoustic factors are associated with membership in each headphone segment. This information can help guide future headphone design that is aimed at a specific class or segment of listener.

9:20

2aEA4. What is the correct ear-canal boundary condition for simulating HRTFs for open-ear listening? Morteza Khaleghi-Meybodi (Facebook Reality Labs, 9845 Willows Rd., Redmond, WA 98052, morteza@fb.com), Pablo Hoffmann, and Tony Miller (Facebook Reality Labs, Redmond, WA)

Emerging open-ear binaural reproduction methods for virtual and augmented reality are an opportunity to rethink how we pose the simulation problem for head related transfer functions (HRTFs). Historically, researchers have experimentally measured HRTFs using a simplifying blocked meatus condition, then used those HRTFs to reproduce spatial audio using low-latency head tracking, an HRTF convolution engine, and carefully calibrated free-air-equivalent coupling over-the-ear headphones. This blocked meatus boundary condition was then carried over as researchers have moved toward numerical simulation of HRTFs, allowing for direct comparison of simulation and experiment. Now that the industry is trending toward numerically-simulated HRTFs and open-ear playback methods, a more realistic ear-canal input impedance needs to be further investigated. In this paper, we report changes in the sound pressure at the entrance to the ear-canal from open to blocked meatus conditions, using a Multiphysics Finite Element model. The blocked meatus boundary is replaced with a nominal frequency-dependent ear-canal input impedance and the pressure-division ratio of open to blocked condition is evaluated. These modeling results suggest that it may not be necessary to perform detailed geometry capture of the ear-canal and that a simplified input impedance is sufficient to simulate the open ear effect.

9:40

2aEA5. Commercializing Piezo-MEMS electroacoustic sensors. Karl Grosh (Dept. of Mech. Eng., Vesper Technologies, 2350 Hayward St., Ann Arbor, MI 48109-2125, grosh@umich.edu)

Microphones are ubiquitous, appearing in consumer electronic devices like mobile phones, smart speakers, and voice activated appliances. In this talk, one path taken to transitioning a particular University-discovered technology, a piezoelectric micro-electro-mechanical-systems microphone, from a research project to a venture backed company (Vesper Technologies) will be described. The low dielectric loss of aluminum nitride (AlN) combined with its elastic and piezoelectric properties proved to be the enabling material for the design of electroacoustic sensors with low input referred noise for in-air acoustic sensing. The availability of industrial quality AlN deposition tools has made the manufacture of these devices at scale for the consumer electronics industry possible. Designs that mitigate the deleterious effects of variable residual stresses and allow for consistent sensitivity over the wafer (from device to device) are described. The flexibility afforded by MEMS technique has enabled the fabrication of various other sensors and experimental examples of function along with modeling results will be shown for both in-air and underwater devices. The potential advantages of enhanced piezoelectric materials will be presented. Mechanisms for leveraging University, Governmental (Local, State, and Federal), and Equity based funding sources are described.

10:00–10:15 Break

10:15

2aEA6. A study on the effect of the vertical sound field control of the flat panel display speaker on audience. SungTae Lee (LG Display,245 LG-ro, Paju-si, Gyunggi-do10845, South Korea , owenlee@lgdisplay.com), Kwanho Park (LG Display, Paju, South Korea), and Hyung Woo Park (IT, Soongsil Univ., Seoul, South Korea)

Technological improvement in the display industry, the organic light-emitting diode(OLED) panel manufacturer changed the quality of picture and sound improvement. In previous research, we introduced the technology that uses the display panel as a speaker and also to improve the sound quality by improving the vibration characteristics and horizontal acoustic characteristics of an OLED panel. In this study, we analyzed the characteristics of the vertical direction sound of the TV and to consider auditory sense at various distances. When watching TV, unlike normal viewers, the infant and child viewers are too close to the screen. Acoustic transmission using conventional TV allows the reflected sound to propagate from downward. So, near the TV and in a low position, a loud sound is formed. This causes hearing damage at close range and low clarity at long range. The exciter speakers can basically eliminate these disadvantages of the sound field. And, by adaptive control of the sound, it is possible to transmit a clear sound in a wide space by reducing loss due to the distance. This paper introduces the results of the study on the improvement of sound quality through control algorithm and the protection of hearing audiences through sound field analysis.

Contributed Papers

10:35

2aEA7. On an improvement of the sound separation ratio according to exciter attachment method of flat TV. Hyung Woo Park (Commun. Eng., Soongsil Univ., 520 Junsangawn, 369 Snagdo-Ro, Dongjak-Gu, Seoul 06978, South Korea, pphw@ssu.ac.kr), SungTae Lee (Commun. Eng., Soongsil Univ., Gyunggi-do, South Korea), Kwanho Park (Commun. Eng., Soongsil Univ., Paju, South Korea), and Myungjin Bae (Commun. Eng., Soongsil Univ., Seoul, South Korea)

In the case of a flat-screen TV, the position of the sound was properly configured by reproducing the sound directly from the left and right sides of the screen. However, focusing on the image quality and design elements, it was hidden above. With several experimental factors, we were able to reproduce a lot of sound from the bottom speaker. This is disadvantageous in that it cannot hear the reproduced sound directly, hears mainly the reflected waves of the space below the space where the information display device is located, and hears different sounds depending on the characteristics of the reflection surface. In this study, we introduce a technique to make a sound with a direct screen that complements these shortcomings. In a large-sized flat-screen TV, the sound drives an exciter attached to the rear, proposing a way to improve forward beam forming depending on the attachment method. Compared to the conventional exciter attachment method, the improved direction ratio is improved by 22% and the voice separation ratio of the multi-channel according to the horizontal view distance is improved. Therefore, this method will contribute well to the use of a big size flat display speaker in multi-channel entertainment applications.

10:50

2aEA8. On listeners preferences in subjective assessment tests of multichannel audio on mobile phones. Fesal Toosy (Elec. Eng., Univ. of Central Punjab, 1 Khayaban e Jinnah, Johar Town, Lahore 54700, Pakistan, fesal@ucp.edu.pk) and Muhammad S. Ehsan (Elec. Eng., Univ. of Central Punjab, Lahore, Pakistan)

ITU standard listening tests like the ITU-R BS.1534-3 (MUSHRA) emphasize on the use of expert listeners especially for rating global attributes like Basic Audio Quality. The ratings of expert listeners are more reliable but naïve listeners are often used for such tests, especially when attributes like Quality of Experience are being measured. Moreover, the results from naïve listeners are actually considered more reflective of average consumer preference. This paper presents further statistical analysis of the results of two such previously conducted tests for assessing multichannel audio versus stereo and mono on mobile phones using headphones. The procedure and materials used in the tests were compliant with the MUSHRA requirements and the test material consisted of audio excerpts with 6 or 8 channels. The multichannel excerpts were object coded and served as the reference which in turn was further down-mixed to different versions of stereo and mono to serve as the medium and low anchors. The media player used had custom HRTF's and rendered the multichannel excepts for a binaural playback on headphones. The analysis involved measure of rank correlations on the results in order to study the listener ratings in more detail. The results showed that for the ratings in both tests, the preference for multichannel, stereo and mono had little correlation with age and gender but a regression model for predicting ratings of one test from the ratings of the other was possible.

11:05

2aEA9. Portable acoustofluidic nebulizer. William Connacher (Mech. and Aerosp. Eng., Univ. of California, San Diego, 4787 Cather Ave., San Diego, CA 92122, wconnach@eng.ucsd.edu), Aditi Jain (Elec. and Comput. Sci., Univ. of California, San Diego, San Diego, CA), Mark Stambaugh, Vincent W. Leung (Qualcomm Inst., Univ., San Diego, CA), and James Friend (Mech. and Aerosp. Eng., Univ. of California, San Diego, La Jolla, CA)

The field of acoustofluidics studies the effects of acoustic waves transferred from piezoelectric solids to small volumes of fluid. The phenomenon of acoustofluidic atomization has been studied for several decades now, and one of its major proposed applications is in pulmonary drug delivery. It is well known that droplets in the 1 to 5 μm range are optimal for lung delivery. High-frequency, oriface-free, acoustic vibration has been shown to produce droplets in narrow size distributions including this range. Until now, this technology has not been practical for point-of-care. We have created the first portable acoustofluidic nebulizer as a proof of concept for this application. This device has been enabled by improved transducer design that can now be driven by standard COTS (commercial off-the-shelf) integrated circuits for efficient frequency-synthesis and power-delivery in the 5-25 MHz, 0.5–2 W range. We use a thickness mode transducer with reduced dimensions and incorporate an on-board resonance search algorithm in order to obtain a true thickness mode acoustic wave. We supply fluid from a wick in contact with the edge of the transducer and promote wetting of the surface via fine abrasion in order to obtain continuous atomization from a thin film. Our device nebulizes 0.1 ml/min of water in the critical 1 to 5 μm diameter range using a standard 3.7 V, rechargeable, lithium-ion battery.

11:20

2aEA10. Ultrasonic inspection of lithium-ion batteries to determine state of charge, state of health, and battery safety. Tyler McGee (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX 78712, tyler.m.mcgee@gmail.com), Erik Archibald, Ofodike A. Ezekoye, and Michael R. Haberman (Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX)

Accurately estimating state of charge (SOC), state of health (SOH), and other metrics of battery safety remains a challenge for lithium ion batteries outside the laboratory, where one may use techniques such as X-ray diffraction, electrochemical impedance spectroscopy, or neutron imaging. Existing *in situ* methods to estimate SOC and SOH employ voltage monitoring or Coulomb-counting. Unfortunately, these methods remain inaccurate because they assume a battery capacity or voltage per discharge, while capacity and voltage are known to vary with C-rate and cycle age. The present work explores the viability of interpreting SOC, SOH, and safety metrics by monitoring and interpreting ultrasonic signals propagating through a battery as it is cycled between charged and discharged states in various environmental conditions. Of specific interest are cases where a battery enters dangerous operating conditions which lead to the formation of gases within the battery

as well as increases in internal temperature, pressure, and battery volume. To investigate this regime, the received ultrasonic signals are monitored over the course of charging and discharging cycles and analyzed in the time and frequency domains in an effort to provide an *in situ* means to estimate SOC, SOH and battery safety metrics.

11:35

2aEA11. Enabling rapid charging lithium metal-based rechargeable batteries through suppression of dendrite growth and ion depletion in the electrolyte via surface acoustic wave-driven mixing. An Huang (Material Sci. and Eng., Univ. of California San Diego, 9500 Gilman Dr., SME Rm. 320, La Jolla, CA 92093, anh081@eng.ucsd.edu), Haodong Liu (Dept. of NanoEng., Univ. of California San Diego, La Jolla, CA), Ofer Manor (Dept. of Chemical Eng., Technion - Israel Inst. of Technol., Haifa, Israel), Ping Liu (Dept. of NanoEng., Univ. of California San Diego, La Jolla, CA), and James Friend (Dept. of Mech. and Aerosp. Eng., Univ. of California San Diego, La Jolla, CA)

Lithium metal is an attractive material for use as anodes in batteries due to its high energy density. However, it is unstable during recharging, with non-uniform Li deposition that leads to porosity, dendrites, rechargeable lithium metal batteries (LMB) have been unrealistic for fifty years with serious safety problems. Over this time, research on chemistry modifications have produced modest improvements, but none that have justified considering LMB over lithium-ion batteries in rechargeable applications. Nonuniform Li deposition during charging occurs due to a Li ion depletion layer adjacent to the anode, especially at high charge rates. By including a small surface acoustic wave device (SAW) into the LMB that produces intense acoustic waves in the electrolyte, rapid submicron boundary layer mixing flow may be generated during charging. This flow largely eliminates the Li ion depletion layer, and because the SAW device is small, solid state, and requires only 10 mW h/cm^2 during battery charging, there is a realistic possibility of incorporating this technology into current batteries under consideration for an electric vehicle, consumer device, and medical applications. The elimination of the ion depletion layer furthermore allows high-rate charging, as we will demonstrate in our electrochemistry and morphological results. The underlying physics will be explained using a closed-form model formed from intermediate asymptotics, and will show the crucial impact of the Peclet number in avoiding the ion depletion layer.

TUESDAY MORNING, 3 DECEMBER 2019 — CAROUSEL, 7:45 A.M. TO 10:50 A.M.

Session 2aEDa

Education in Acoustics, Student Council, and Women in Acoustics: Mentoring Graduate and Undergraduate Students

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, Pennsylvania 16802

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

Chair's Introduction—7:45

Invited Papers

7:50

2aEDa1. Information and suggestions for new mentors of research students. David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109-2133, drd@umich.edu)

Identifying, attracting, motivating, and mentoring research students (mentees) are challenging tasks for junior faculty members and research scientists (mentors) who are typically also concerned with obtaining research funding, teaching well, and/or impressing colleagues. This presentation covers three general concepts for effective mentoring that have worked well for the author during nearly 30 years as a faculty member. (1) Let the truth be your only story. In addition to its moral clarity, this policy increases research efficiency and tends to be respected even when it's unpopular. (2) Know who students are and what motivates them. Students are academic free agents with a variety of motivations who commonly bring human-interaction complexities to the mentor-mentee relationship. (3) Engage in student-centric advising. The mentor acts as an academic parent who (typically) determines the mentee's first research steps, guides their subsequent development, and then supports them through their final degree. Descriptions, explanations, and examples related to these three concepts are presented. Although this content is primarily derived from the author's academic experience advising graduate students, it may be useful in industry and government settings as well, and for mentors and mentees alike, whenever extended duration advising requires the development of a mentor-mentee relationship.

8:05

2aEDa2. Mentoring the person: Building on strengths and managing challenges. Barbara Shinn-Cunningham (Neurosci. Inst., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, shinncns@gmail.com)

Mentoring is not about setting up expectations of perfection, but working with each person to help them succeed. Every person, from undergrad to senior professor, has some skills that come naturally, and some that, …well…, do not. This talk will reflect on the fact that good mentoring involves helping each unique person identify both their "super powers"—as well as their weaknesses. Each person has things that they are good at, that they enjoy, and that speak to their values (their super powers). But knowing how to mitigate weaknesses is equally important, whether it is through recruiting a good collaborator to fill in a gap, working to strengthen a particular skill, or simply forging a path where a particular weakness is not critical. Along some paths, perseverance and an ability to see the big picture may be more critical than programming skills or mathematical wizardry; along others, raw intellectual horsepower and an ability to focus may be key, without relying on having to work closely with others. Without naming names, I will provide some examples of how some of my mentees have succeeded by building on their strengths while managing their personal and professional challenges.

8:20

2aEDa3. Choose the right people and help them realize their worth. Christy K. Holland (Internal Medicine, Div. of Cardiovascular Health and Disease, and Biomedical Eng., Univ. of Cincinnati , Cardiovascular Ctr. Rm. 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu)

Students flock to "good" labs, places where the love of science first strikes. To attract talented people, your lab must be a place that breeds good science and successful trainees. Hallmarks of a wonderful lab include exciting projects, collaborative publications, and students and postdoctoral fellows who have found good jobs. The head of the lab knows how to and is willing to pass along these skills to other members of the lab. Strategies for how to choose your people carefully and how to determine when to hire a technician, lab assistant, student, or postdoctoral fellow will be outlined. Examples of how to determine a candidate's capability of performing the job, and if the candidate will fit into the existing lab will be discussed.

8:35

2aEDa4. Building and maintaining an active research laboratory with undergraduates. Laura Kloepper (Biology, Saint Mary's College, 262 Sci. Hall, Notre Dame, IN 46556, lkloepper@saintmarys.edu)

Immersing undergraduates in research is important for the advancement of students into graduate school and the workforce. Mentoring undergraduates, however, often requires a different approach than mentoring graduate students. In this talk, I will share practical tips on how I have built and maintained an active, funded research laboratory at a primarily undergraduate institution. Specifically, I will explain my strategies for attracting and retaining students, my tiered mentoring approach for first-year through senior students, and the benefits I have received from these mentoring relationships.

8:50

2aEDa5. Reflections of mentoring midshipmen and high school students at the United States Naval Academy—With student recollections. Emily V. Santos (Dept. of Psych., Univ. of Maryland, Baltimore County, 1000 Hilltop Circle, Annapolis, MD), Katherine A. Haas (Dept. of Bio-Chemistry, St. Mary's College of Maryland, 47645 College Dr., St. Mary's City, MD), Jenna M. Cartron (Dept. of Mater. Sci. and Eng., Johns Hopkins Univ., 3400 N. Charles St., Laurel, MD), and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

For 38 years, MSK has developed mentoring undergraduate students in the Physics Department. Mentorship is a unique dance, a growing tree—with kind branches embracing any challenge—a give and take enjoyed by both the mentor and mentee. It should be fun—filled with goals to achieve (often wrapped in a research challenge). "Acoustics" and "Underwater Acoustics and Sonar," courses—inspire projects. Students have presented research at ASA meetings. Some become co-authors on published papers. For 35 years, MSK developed a high school mentorship program at USNA. Rising juniors and seniors may apply. HS mentees start their research on a project that evolves over one or two months working twice a week for 2.5–3 h sessions. Students often continue over the summer. Recently, MSK took HS mentorship students to the ASA Boston Meeting, the 17th ISNA Meeting (Santa Fe 2018), Louisville and Internoise 2019 in Madrid. Three HS seniors at the ISNA conference presented research (published in POMA) on (1) nonlinear acoustic landmine detection with wetted and dry beads [EVS], (2) acoustic levitation and nonlinear sound scattering by a bubble [KAH], and (3) the interaction of two crossed ultrasonic streaming jets [JMC]. Their dedication and insights are highly valued.

9:05

2aEDa6. Mentoring when you don't have time to mentor. Scott D. Sommerfeldt (Dept. of Phys., Brigham Young Univ., N249 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

In my position as a faculty member at Brigham Young University (BYU), I served for ten years as the dean of my college. Mentoring of students is a highly valued activity at BYU, so I had reasons to promote strong mentoring in the college, as well as to seek to mentor students myself—both graduate and undergraduate. However, there really is not any time in the job description to devote towards mentoring, so how does one try to mentor effectively? This paper will review some of my experiences, as well as principles I learned while in the dean's office to help me try to be an effective mentor—principles that can actually extend to all mentoring situations. I will also share some initiatives that were undertaken by the college to try and promote better mentoring throughout the college.

9:20

2aEDa7. Ideas for improving each phase of a mentored research relationship. Kent L. Gee (Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

A challenging part of an academic's job is developing productive research relationships as a mentor. This paper presents ideas for improving a mentor's effectiveness at different stages, from student recruiting through the relationship's redefinition upon graduation. These ideas stem both from personal experiences with undergraduate and graduate students and from a review of relevant literature.

9:35

2aEDa8. Mentoring with intent, even when you aren't a student's research advisor. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

The roles of mentoring and advising are closely related, but not necessarily equivalent, and the distinction between the two is often blurred. It is most often expected that a student's research advisor will also be their primary mentor. The research literature exploring how graduate students learn (especially the theory of "Community/Landscapes of Practice") suggests that most learning happens outside of the classroom, that students need a network of mentors, and that a student's research advisor should not play all of the mentoring roles for that student. I was quite surprised to receive the 2016 ASA Student Council Mentor Award, because in my current role as a teaching professor at Penn State, I do not advise many graduate students earning their M.S. or Ph.D. who do their research under my direct supervision. However, I do play an active role as a mentor, guiding the academic and professional development of many students in our program. I played a similar role to a large number of undergraduate students at my previous institution. In this talk, I will discuss several ways that faculty can mentor to students who are not their own research advisees.

Contributed Papers

9:50

2aEDa9. Research group meetings as a professional development book club. John R. Buck (ECE, UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, jbuck@umassd.edu)

Many of the soft skills important for professional success in a research career are not taught in classes listed in the graduate course catalog. This talk describes the professional development goals and practices in the weekly meetings of the Signal Processing Group at the University of Massachusetts Dartmouth. Meetings in alternate weeks switch between students discussing readings from books on important professional skills and students presenting practice conference talks. For the first semester, we read and discuss a book on scientific presentations, most recently Allen's "The Craft of Scientific Presentations." Book choices for the second semester in recent years include "Writing Science" by Schimel, "The Visual Display of Quantitative Information" by Tufte, and "The Signal and the Noise" by Silver. Every student gives a practice presentation every semester, and receive constructive peer and advisor critiques. The critiques are based on the framework of the book on scientific presentations we read together during the first semester. As a result, students have presented several times before an audience of their peers before their first professional conference presentations.

10:05

2aEDa10. Here, go read these papers: An approach to engage undergraduate and graduate research students with literature. Teresa J. Ryan (Dept. of Eng., East Carolina Univ., 248 Slay Hall, Greenville, NC 27858-4353, ryante@ecu.edu) and Melissa A. Hall (Mech. Eng., Univ. of Cincinnati, Cincinnati, OH)

One of the key tasks of a research mentor is to foster the skill of independent inquiry. Central to effective independent inquiry is the ability of students to extract meaningful value from the topical research literature. For several years, students applying to be part of the acoustics and vibrations lab at East Carolina University have been required to summarize a select research paper as part of the interview process. That task provides a valuable barometer to evaluate prospective research students. It also has provided the nucleus for an evolving system to mentor students through the process of learning how to effectively interact with topical literature. This system has been used both to onboard new research students and to support directed research and independent study courses. This talk will present the approach—implemented in an asynchronous and largely self-paced manner with the university learning management system—and some key case study examples.

10:20–10:50
Panel Discussion

Session 2aEDb

Education in Acoustics: Acoustics Education Prize Lecture

Marcia J. Isakson, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, Texas 78712

Thomas G. Muir, Cochair
Applied Research Laboratories, University of Texas at Austin, P.O. Box 8029, Austin, Texas 78759

Chair's Introduction—11:00

Invited Paper

11:05

2aEDb1. Learning and teaching acoustics through bubbles. Preston S. Wilson (Mech. Eng., Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712-0292, pswilson@mail.utexas.edu)

By a lark, I started my acoustics career as a graduate student studying a single big bubble in the form of the combustive sound source. By at least the luck of being in a certain place at a certain time, I finished my graduate education by studying bubbles in the form of the acoustics of bubbly liquids. It turns out that the subject of bubbles provides an excellent lens through which to learn and teach acoustics. One starts with lumped element approximations which lead to simple harmonic oscillators and resonance. You can study the bubble oscillator as linear system and encounter increasingly complex nonlinear behavior at comparatively low amplitudes. A bubble is a source of sound initially modeled as a simple spherical source and can of course behave as a spherical scatterer. Add one or many bubbles and one can study more complex scattering and propagation that includes dispersion and loss. Along the way one encounters effective medium theories of varying complexity and statistical descriptions of acoustic phenomena. There are also applications that require inference and inverse techniques. You can even apply a model of a single bubble to study the expansion and contraction of the universe. Examples of these including demos and videos will be presented.

Session 2aMU

Musical Acoustics: Experimental Methods in Musical Acoustics: Best Practices

Andrew A. Piacsek, Chair
Physics, Central Washington University, 400 E. University Way, Ellensburg, Washington 78926

Chair's Introduction—8:00

Invited Papers

8:05

2aMU1. Silk purse? Brass-instrument research with limited resources. Robert W. Pyle (Stephens Horns, 11 Holworthy Pl., Cambridge, MA 02138, rpyle@icloud.com)

Can we make a silk purse from the proverbial sow's ear? That is, can we do meaningful experiments with inexpensive equipment? It is certainly much easier than it used to be. The advent of digital audio and cheap computing power has revolutionized musical acoustics research. In this paper the author looks at several projects on brass instruments, most of which he funded out of his own pocket. Some yielded better results than others, but all pointed toward interesting work yet to be done.

8:25

2aMU2. To infinity and beyond: The life story of brass instrument shock waves. D. Murray Campbell (School of Phys. and Astronomy, Univ. of Edinburgh, James Clerk Maxwell Bldg., Mayfield Rd., Edinburgh EH9 3JZ, United Kingdom, d.m.campbell@ed.ac.uk)

One of the most spectacular features of the sound of trumpets, trombones, and french horns is the way in which the spectral content evolves during a crescendo. The gentle rounded timbre which these instrument create when played very quietly becomes increasingly bright as the loudness increases, and at the fortissimo level the sound develops a snarling brilliance described as "brassy." In 1980, Jim Beauchamp suggested that nonlinear sound propagation might play a role in this spectral enrichment, and in 1996, experiments by Mico Hirschberg, Joel Gilbert and colleagues confirmed that the cumulative effect of nonlinear distortion in a trombone could lead to shock wave formation. This paper reviews the experimental studies which have subsequently been undertaken to understand better the circumstances which lead to the theoretically infinite pressure gradient of a shock wave, and the way in which the wavefront evolves as it travels on to the bell and into the open space beyond.

8:45

2aMU3. High precision measurements of acoustic impedance: Applications to the voice and respiratory tract, musical instruments and their players. Noel N. Hanna (UNSW Global, 223 Anzac Parade, Kensington, New South Wales 2033, Australia, n.hanna@unsw.edu.au), John Smith, and Joe Wolfe (School of Phys., UNSW, Sydney, New South Wales, Australia)

The acoustic impedance spectrum of the vocal tract and trachea is important in speech and singing. The operation of musical wind instruments depends on the impedance spectra of their bores, and sometimes also on the impedance spectra of the player's vocal tract. Here, we describe two measurement techniques. The three microphone technique, calibrated with three non-resonant loads, achieves large dynamic range measurements of the frequencies, magnitudes, and bandwidths of resonant duct maxima and minima. The impedance minima are particularly important in flute family instruments and in the voice, where resonances correspond to minima at the lips. These measurements can also be used to estimate the duct geometry and acoustic properties at the distant end, which may be otherwise inaccessible. The capillary or current source technique has been used to measure impedance maxima inside the mouths of wind players during performance, e.g., where saxophonists tune their tracts to the note played, and for "ecological" measurements during speech and singing that give estimates of resonant frequencies that are rather better than those from formant estimation at high pitch. This technique has recently been updated to run in real-time and made available as open source software, with modest equipment requirements.

9:05

2aMU4. Measuring free reed oscillation: Some case studies. James P. Cottingham (Phys., Coe College, 1220 First Ave. NE, Cedar Rapids, IA 52402, jcotting@coe.edu)

This paper presents an overview of efforts to measure several quantities related to sound production in free reed instruments. All were undertaken with limited financial resources for equipment and a professor who started with limited experience in experimental physics, but with a good supply of talented and creative undergraduate students. The case studies to be presented will include detecting the presence of higher transverse modes and torsional modes of vibration in an air-driven free reed, detection and significance of these modes in the initial transient, measurement of the volume airflow as a function of time in an oscillating free reed, and using a combination of measurement and theoretical calculation to predict the sounding frequencies of free reed pipes with non-uniform bore shapes.

9:25

2aMU5. Comprehensive experimental characterization of violin vibration and radiation dynamics. George Bissinger (Phys. Dept., East Carolina Univ., Greenville, NC 27858, bissingerg@ecu.edu)

Measuring *only* violin radiation cannot remove ambiguities in resonance peak identification arising from various minor substructure couplings, or whether the radiation arose from the *f*-holes or surfaces. Measuring *only* vibrations gives no clue as to how well each mode radiates or even how the vibrational energy was dissipated. To put these into a structural acoustics milieu, comprehensive measurements of all the violin's energy loss paths were essential. To accomplish this a wide range of calibrated transducers (accelerometers, multiple microphones, near- and far-field microphone arrays, scanning laser vibrometers) were used in a wide range of experiments: cavity mode analysis using interior microphones; interior gas-exchange; EMA on a violin with and without a soundpost; waterfill experiments on a rigid violin-shaped cavity; automated zero-mass-loading, surface-normal (and 3-D) scanning laser EMA on violins supported in a near-zero-damping support fixture; automated far-field radiation measurements over a sphere in an anechoic chamber; bridge tuning effects; near-field acoustical holography over the ***f***-holes. Each technique had limitations, some severe. Some experiments were "table top"; some required expensive associated facilities like an anechoic chamber. Costs of transducers and power-amplifier-control electronics, computer-based acquisition hardware/software varied dramatically; most operational problems occurred in the computer-based data acquisition/analysis systems.

9:45

2aMU6. Experimental characterization of basic violin structural acoustics. George Bissinger (Phys. Dept., East Carolina Univ., Greenville, NC 27858, bissingerg@ecu.edu)

Considering its simple construction the complex material-shape traditional violin, fashioned from nominally orthotropic materials to form a ported, compliant-wall cavity, enjoys a notable scientific complexity, e.g., four separate radiation mechanisms, coupling between the two lowest cavity modes A0 (Helmholtz resonance, ~275 Hz) and A1 (1st longitudinal "slosh" mode, ~470 Hz), and between A0 and two 1st corpus bending (B1) modes near 500 Hz. Comprehensive EMA (velocity/force), radiativity (pressure/force) and near-field acoustical holography (***f***-hole volume flows/force) measurements plus bridge tuning systematics were combined to establish a *dual*-region structural acoustics model of the violin. In the "deterministic" region the large-volume-change B1 modes drive large, in-phase *f*-hole volume flows accounting for >50% of B1 radiation *and* excite A0; oppositely directed upper-bout/lower-bout *cavity* volume flows excite A1. A wall-driven, dual-Helmholtz resonator network successfully modeled A0 and B1 radiation. In the "statistical" region where the major bridge and plate tuning effects occur, radiation efficiency (averaged-over-sphere radiativity divided by averaged-over-corpus mobility) led to radiation damping and critical frequency. With total damping from mobility fits the radiation/total damping ratio (fraction of vibrational energy radiated) was combined with a parametric model of mobility. Splicing the regions created the "dynamic filter" model of violin radiation.

10:05–10:20 Break

10:20

2aMU7. Electronic speckle pattern interferometry: A powerful but underutilized experimental technique. Thomas Moore (Dept. of Phys., Rollins College, 1000 Holt Ave., Winter Park, FL 32789, tmoore@rollins.edu)

Electronic speckle pattern interferometry is a cost effective technique for imaging the deflection shapes of harmonically vibrating objects. The images produced by the interferometer show the vibrating object superimposed with fringes representing contours of equal out of plane amplitude. These images can be used to determine deflection shapes, find and characterize resonances, and ensure the proper placement of sensors. Constructing the interferometer requires only minimal optical expertise, and with slight modifications the interferometer can be made to image static displacement, transient vibrations, fluid flow and thermal effects.

10:40

2aMU8. Electronic speckle pattern interferometry: applications to the structural acoustics of percussion instruments. Randy Worland (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Electronic speckle pattern interferometry (ESPI) provides real-time images of operating deflection shapes of acoustically driven structures. This technique is particularly useful in the study of musical percussion instruments containing bars, plates, and membranes. Musically relevant examples involving drumheads and vibraphone bars are described. In addition, a dual-ESPI system that produces simultaneous measurements of two surfaces of an object (e.g., the two heads of a drum) is illustrated. Imaging through water is also shown, as applied to the musical technique in which an orchestral crotale is submerged in water to varying depths. Finally, the interferometer is used to image the confined modes of the "musical saw" when bent into its characteristic "S"-shape. Practical issues of object mounting and image production are discussed.

11:00

2aMU9. Musical instrument directivity measurements. Timothy W. Leishman (Phys. and Astronomy, Brigham Young Univ., N247 ESC, Provo, UT 84602, twleishman@byu.edu) and Samuel D. Bellows (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Over the past several decades, researchers have developed various techniques to measure directional radiation patterns of musical instruments. After summarizing and categorizing several of them, this presentation will focus on recent developments made by the authors to measure directivities with high angular resolution. The frequency-dependent patterns of 16 played musical instruments were acquired using 2522 unique microphone positions over a sphere in an anechoic chamber. This led to associated frequency-dependent spherical-harmonic expansions with readily shareable coefficients. Several steps were implemented to improve and validate the measurements, and make their costs manageable. The presentation will discuss key results and future directions.

Contributed Papers

11:20

2aMU10. Directivity of the musical instruments and voice in a real situation using beamforming techniques. Mojtaba Navvab (Architecture, Univ. of Michigan, 2000 Bonisteel Blvd, Art and Architecture Bldg., Ann Arbor, MI 48109-2069, moji@umich.edu) and Andy Meyer (GFai Tech GmbH, Berlin, Germany)

Musical instruments are considered natural sound sources with specific directivities. Each instrument unique directivity needs to be identified and implemented in auralization of sound in room acoustic designs. This paper shows detailed directivity measurements on selected musical instruments, human voice, and impulse sources used in room acoustics design under realistic conditions. This work presents results that are measured using acoustic beamforming techniques. Measurements are based on simultaneous recording using a hemispherical array of 120 microphones. The directivity data are measured and analyzed in the time, and frequency domain, and effects in the near field of musical instruments are explored. Dynamic variations in the musical instruments' directivity relative to sound power, loudness (intensity, pressure), projection, and the apparent size of the sound or volume are presented. Recorded data are presented in static or dynamic mode as well as in both 2- and 3-dimensional format using the beamforming software as well as Math lab algorithm. The results of this work are applicable toward improvement in the quality of auralization of musical instrument in simulation tools and vocal production studies and room acoustic sound filed. Evaluation of their projected radiations' pattern contributes to a more realistic acoustic design simulations or virtual reality applications.

11:35

2aMU11. Speckle pattern interferometric studies of guitar top plate materials. Spencer Thulin (Phys., Whitman College, 345 Boyer Ave., Walla Walla, WA 99362, thulinse@whitman.edu), Sam Tabbutt (Phys., Whitman College, Seattle, WA), and Kurt Hoffman (Phys., Whitman College, Walla Walla, WA)

We present experimental measurements of the resonances of guitar top plates using a speckle pattern interferometer. The studies include sanded spruce starting materials of different grades and finished commercial guitars. The spruce top plates consist of three different wood grades reflecting the quality of the wood grain. The normal mode frequencies and patterns are measured without sides to be consistent with the tap tones a Luthier would use to determine the quality of the top plate. We will discuss the ways in which the wood grade changes the frequency and distribution of normal mode frequencies due to variations in the grains of the raw wood. We will also compare these results to measurements on commercial guitar top plates.

TUESDAY MORNING, 3 DECEMBER 2019 CRYSTAL, 8:30 A.M. TO 10:30 A.M.

Session 2aNSa

Noise, Animal Bioacoustics, and Underwater Acoustics: Effects on People and Wildlife from Transportation Noise (Land, Air, Sea), as well as Innovative Solutions for Reducing Noise

James E. Phillips, Cochair
Wilson Ihrig, 6001 Shellmound St., Suite 400, Emeryville, California 94608

Bonnie Schnitta, Cochair
SoundSense, LLC, 46 Newtown Lane, Suite One, East Hampton, New York 11975

Chair's Introduction—8:30

Invited Papers

8:35

2aNSa1. Estimating propagation and audibility of industrial noise in subnivean polar bear dens. Megan A. Owen, Anthony M. Pagano (San Diego Zoo Global, San Diego, CA), Wisdom Sheyna (Fairweather Sci., Anchorage, AK), and Ann Bowles (Hubbs-SeaWorld Res. Inst., 2595 Ingraham St., San Diego, CA 92109, abowles@hswri.org)

Polar bear (*Ursus maritimus*) maternal dens are ephemeral and difficult to monitor, so predictive models of noise penetration into dens are needed. Noise propagation into artificial snow dens was monitored from 9 sources typical of industrial activities (2 aircraft, 2 over-tundra tracked vehicles, 4 wheeled on-road vehicles, and humans walking) near Milne Point, Alaska. Dens were built in 4 configurations, 2 depths (100 cm, 70 cm) and 2 closure conditions (closed, open) to model variability in den roof thickness and breeding stage (before, after emergence). An existing polar bear audiogram was used to predict detection probabilities. Levels <200 Hz would be heard poorly; after accounting for hearing, aircraft had $\geq 75\%$ probability of being detected within dens at distances <1600 m, and ground-based sources at distances <800 m. On average, closed dens reduced noise levels by 15 dB (re 20 μPa) relative to open dens, but there was interaction between den closure and depth. Some sources were likely to be detected farther from dens than expected. The results reinforce the importance of maintaining disturbance buffer zones around polar bear dens.

8:55

2aNSa2. The effect of transportation noise on sleep and its consequential effect on health. Jo Solet (Harvard Med. School and Cambridge Health Alliance, 15 Berkeley St., Cambridge, MA 02138, joanne_solet@hms.harvard.edu) and Bonnie Schnitta (SoundSense, LLC, Wainscott, NY)

Sleep disruption by noise is a well-known phenomenon that has demonstrated negative health and cognitive impacts. Furthermore, it has now been determined that individuals suffering from certain health conditions, including PTSD and atrial fibrillation, may actually experience enhanced noise sensitivity with additional risks. This paper presents the results of several studies that confirm the importance of addressing transportation noise, over which individuals typically have no control, to protect environments intended for sleep such as hospitals, senior housing facilities, and residences.

9:15

2aNSa3. Calculating and specifying acoustical performance of exterior façade assemblies. John LoVerde (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com) and David W. Dong (Veneklasen Assoc., Santa Monica, CA)

The acoustical performance of the exterior façade determines the noise exposure of residents to transportation noise. Façade assemblies with high acoustical performance have a large impact on construction in terms of cost, dimension, and schedule. Uncertainties in the acoustical performance require safety factors and generally conservative design in order to ensure that the requirements are met, with corresponding increase in impact to the project. Historically, STC ratings have been used to describe the performance of exterior façade assemblies. A variety of alternative ratings and calculation methods have been proposed to improve the accuracy or precision of the description. Examples include OITC and the spectrum adaptation terms of Rw defined in ISO 717-1. The authors have previously analyzed the use of OITC versus STC for describing the sound isolation of exterior façade assemblies [J. LoVerde and W. Dong, "Comparison of sound transmission class and outdoor-indoor transmission class for specification of exterior facade assemblies," *J. Acoust. Soc. Am.* **141**, 3539 (2017)]. This analysis is extended to additional ratings and methods with the goal of minimizing the uncertainty of the analysis. Recommendations for calculation and specification methods of the acoustical performance of exterior façade assemblies are presented.

9:35

2aNSa4. Traffic noise in street-level hotel rooms. Sarah Taubitz (45dB Acoust., LLC, PO Box 12275, Denver, CO 80212, st@45db.com)

A one-story butler-style building was renovated to become a trendy 6 guest-room hotel, at the corner of a city intersection. Road noise infiltrates the rooms, some guests have publicly complained. Various measurements were made and will be presented, including: Noise Reduction (NR); Reverberation Time (RT); short- and long-duration overall and frequency-band LAeq; AOITC; and leakage path investigations. Several noise treatments will be presented.

9:55

2aNSa5. Case study review of vibration mitigation for a building foundation isolation using elastic polyurethane bearings. Jessica Scarlett (Getzner USA, 8720 Red Oak Blvd, Ste. 400, Charlotte, NC 28217, jjstamey@gmail.com) and Alexander C. Born (Getzner USA, Charlotte, NC)

This abstract addresses a case study review of a new construction mixed-used building located in New York City, New York. The building site was located adjacent to an existing subway tunnel. Measured vibrations from train pass-by events were perceptible and above Federal Transit Administration "Daytime Criteria." Custom elastic polyurethane bearings were designed to provide vibration and ground-borne noise mitigation according to specified insertion loss values. The elastic bearings were installed at the foundation level along with additional elastic structural components. Post-construction vibration measurements were compared to initial site measurements and theoretical calculations of the implemented solution. Accuracy of the theoretical model to finalized isolated structure were analyzed.

Contributed Paper

10:15

2aNSa6. Doppler's shift on aircraft noise propagation in modern aircraft noise prediction tools. Yiming Wang (Purdue Univ., 2120 McCormick Rd., Apt. 711, West Lafayette, IN 47906, wymchihiro@gmail.com) and Kai Ming Li (Purdue Univ., West Lafayette, IN)

Doppler's effect is an important factor in the prediction of the noise levels caused by en-route aircrafts. This study is focused on the influence of Doppler's effect on the propagation of aircraft noise. The received sound pressure can be increased/decreased by Doppler's effect depending on the relative location of the sound source and receiver. At the same time, the spectral shape of the noise is shifted by this effect which has an impact on the measured A-weighted noise levels. This influence is largely due to the fact that A-weighted correction factors of mid-frequency range are much larger than that those in the low-frequency regime. In this paper, a series of field trial was used to study the propagation effect of noise generated by a propeller-driven aircraft. Aviation Environment Design Tool (AEDT), a widely accepted noise prediction model of aircraft noise, is used to compare with the predictions according to a ray-based numerical model. The inclusion of Doppler's effect in the noise prediction model has shown to have a better agreement with the field measurement results. The influence of Doppler's effect on the noise-power-distance (NPD) curves of several different aircrafts is also presented. [Work sponsored by ASCENT, FAA Center of Excellence for Alternative Jet Fuels and Environment.]

Session 2aNSb

Noise: Outdoor Entertainment Noise

David Manley, Cochair
DLR Group, 6457 Frances St., Omaha, Nebraska, United States

K. Anthony Hoover, Cochair
McKay Conant Hoover, 5655 Lindero Canyon Road, Suite 325, Westlake Village, California 68133

Chair's Introduction—8:30

Invited Papers

8:35

2aNSb1. Noise in a public square. Sergio Beristain (IMA, ESIME, IPN, P.O. Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

Nowadays many public squares have been hosting almost every day many recreational activities, some of them previously advertised, but some not, ranging from single performers with an acoustic instrument, small dancing groups or theatrical performances, up to electronic amplified musical events, either for free or for a fee. This kind of events can be sponsored by the musicians or performers themselves, broadcast or recording companies as promotionals, or by the local and federal governments for political purposes, which for the people living around the square is a nice and free entertainment, but after a few events, together with the implementation processes become a frequent annoyance for their normal living or working activities. In many instances sound levels are well beyond any standard, and due to its variable nature, very hard to control. A case is presented, showing levels, events duration, together with approval and disapproval comments.

8:55

2aNSb2. Community noise monitoring around Universal Studios Hollywood. Randy Waldeck (CSDA Design Group, 475 Sansome St., Ste. 800, San Francisco, CA 94111, rwaldeck@csdadesigngroup.com)

As part of the development agreement/Specific Plan for Universal Studios Hollywood, community noise monitoring is required to maintain acceptable Studio-generated noise levels in the neighborhoods surrounding Universal Studios. Universal Studios operates 24 h a day, operating as both a theme park hosting 9M guests/year and an active film/TV studio covering over 400 acres. Universal Studios' activities generate many different types of noise: crowd screams/cheers, percussive/explosive events as part of filming and the theme park attractions, music from the attractions and filming, and operational noise such as landscaping, set construction, etc. The community noise monitoring is conducted four to six times a year at six residential locations closest to Universal Studios. The monitoring occurs for 24 (continuous) hours, and each monitoring location is staffed for the entire duration so that each noisy event can be noted, characterized, and its source determined (Universal Studios or other non-Universal source). The results of the noise monitoring are compared against the comprehensive Specific Plan noise criteria and a determination of compliance is made. Based on the measurement results, adjustments/modifications to Universal Studios operations, equipment, sound systems, etc. are made to maintain compliance. A discussion of the activities and noise sources, measurement program, and lessons learned will be discussed.

9:15

2aNSb3. Moving ultra. Bennett M. Brooks (Brooks Acoust. Corp., 49 N. Federal Hwy., Unit 121, Pompano Beach, FL 33062, bbrooks@brooksacoustics.com)

The Ultra Music Festival is an international electronic dance music (EDM) franchise. Ultra started in Miami Beach but has been in downtown Miami for over a decade. In 2018, the attendance at the 3-day Ultra fest in the public Bayfront Park was about 200K. The venue is surrounded by high-rise apartment buildings. During the week of sound checks prior to the event, many residents are seen pulling suitcases to taxis/ubers to escape the onslaught. The buildings literally shake. A coalition of residents in downtown Miami sought to monitor the event and to reduce its negative impact on the neighborhood. Bass levels (40 to 125 Hz) approached 120 dB at an adjacent residence. In this frequency range, the Cavanaugh criteria metric (L1—L90) for the impact of outdoor entertainment facilities on residential neighbors was close to 50 dB. The Miami City Commission voted to not renew the Ultra contract for Bayfront Park. In 2019, the festival was moved to the nearby island of Virginia Key, where it generated numerous complaints. As of this writing, Ultra is in political limbo, with its future location yet to be determined. Measured sound data from the event and Ultra's current status will be presented.

9:35

2aNSb4. Simple solutions, happy clients. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

Various problems involving outdoor entertainment noise were resolved with simple solutions, which were especially gratifying when clients had presumed expensive and complicated solutions. This paper will review some examples, including a lakeside amusement park ride largely intended to generate group screams at regular intervals, an outdoor town center featuring evening movies and weekend performances, an NFL stadium that hosts several rock concerts every year, and Mumbai "firecracker" bands impacting Bollywood soundstages.

9:55

2aNSb5. What our prehistoric ancestors can teach us about echoing soundscapes that is relevant to modern noise studies. Steven J. Waller (Rock Art Acoust., 1952 Sonoma Ln., Lemon Grove, CA 91945, wallersj@yahoo.com)

Results of archaeoacoustic studies are applied to outdoor entertainment noise investigations. When considering impacts of noise in modern urban environments, it is well to consider the evidence left by our ancestors regarding the influence of reflective sound surfaces, as well as cultural perceptions of the sounds themselves. Acoustic research into the placement of global prehistoric rock art has revealed that our ancestors selectively chose echoing environments to produce cave paintings and canyon petroglyphs. Furthermore, those images depicted often match descriptions of echo spirits contained in ancient echo myths from around the world. The cultural perception of echoes was that these "extra" sounds were considered mysterious answers from sacred beings worthy of worship—the same phenomena that today are understood as sound wave reflections and considered annoying. Implications for modern noise investigations are discussed. Measure not just noise source amplitude, but consider also the effect of reflective urban structures to multiply, distort and overlap noises, sometimes acting as whisper galleries to cause sound to carry far down "urban canyons." Also, relative to perception, a given dB level can affect different listeners in different ways, e.g., certain music perceived as entertaining to some may be perceived as unpleasantly disturbing to others.

10:15–10:30 Break

10:30

2aNSb6. Noise predictions of sound systems using system data and complex summation. Nick Malgieri (EAS, d&b audiotechnilk, 30A Rosscraggon Rd., Asheville, NC 28704, nick.malgieri@dbaudio.com)

As the number and size of amplified outdoor events in urban areas is increasing, so are the challenges with accompanied noise immissions and therefore the need for accurate noise predictions. An advanced method for predictions of sound systems is introduced, that uses specific system data and applies complex summation. The system data, including all acoustically relevant characteristics, such as positioning, orientation and directivity of loudspeakers, delays and electronic filters, are simply imported with a system design file already in use by the system provider (d&b's ArrayCalc simulation software). This procedure eliminates any inaccuracies and inefficiencies because it spares the process of re-modeling a sound system in the noise prediction software and ensures predictions using the actual system design. Also, and with increasing importance, advanced electronic filtering (e.g., IIR, FIR) is automatically included. Environmental noise modelling software usually does not consider complex summation between sources because they are assumed to be un-correlated with one another (as in traffic and industry). However, in sound systems, the signals sent to the loudspeakers are mostly correlated. Furthermore, modern line-array systems use coherence effects to further tailor directivity. Therefore, the sound propagation calculation was extended for complex summation and the method was implemented in SoundPLAN software and in NoizCalc (d&b audiotechnik).

10:50

2aNSb7. Detecting instances of focused crowd involvement at recreational events. Mylan R. Cook (Phys. and Astronomy, Brigham Young Univ., N201 ESC, Provo, UT 84602, mylan.cook@gmail.com), Eric Todd (Brigham Young Univ., Provo, UT), David S. Woolworth (Roland, Woolworth, and Assoc., Oxford, MS), Kent L. Gee, and Mark K. Transtrum (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

This paper describes the development of an automated classification system for detecting the amount of focused effort present in crowd cheering. The purpose of this classification system is for situations where crowds are to be rewarded for not just the loudness of cheering, but for a concentrated effort, such as in Mardi Gras parades to attract bead throws or during critical moments in sports matches. It is therefore essential to separate non-crowd noise, general crowd noise, and focused crowd cheering efforts from one another. The importance of various features—both spectral and low-level audio processing features—are investigated. Data from both sporting events and parades are used for comparison of noise from different venues. This research builds upon previous clustering analyses of crowd noise from collegiate basketball games, using hierarchical clustering with both supervised and unsupervised machine learning approaches.

11:10

2aNSb8. Comparison of SoundPlan 4.0 noise modeling software and field noise measurements results at two different outdoor entertainment spaces (Glamping and Winery venues). Juan C. Montoya (none, 41 Churchill St., Springfield, MA 01108, jmontoya@csacoustics.com)

Evaluation of noise in outdoor entertainment spaces is a topic of interest in our communities . Noise from outdoor entertainment spaces has turned into an unwanted activity in rural communities due to high levels sound pressure from music events and crowd noise. There are numerous complaints from neighbors or neighborhood associations pertaining noise pollution around areas near these spaces.

For this case study, field noise measurements were conducted in a winery for a regular concert season. Evaluation of the noise predictions using SoundPlan 4.0 for the winery venue and for a Glamping ("luxurious camping") venue are the main focus of this study. The paper aims to determine a correlation between field noise measurements, methodology from ISO 9613-2:1996 Standard "Attenuation of sound during propagation outdoors," and noise prediction results using SoundPlan 4.0 computer modeling software.

11:30

2aNSb9. Amplified concert noise modeling with environmental noise software. David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com) and Joseph Horesco (Acentech, Trevose, PA)

The transformation of an existing park platform stage and 0.5 acre listening lawn into a modern outdoor pavilion and amphitheater with 4 acres listening area presents potentially significant challenges for noise impact changes to the surrounding community. Previous site studies had been completed but new designs and marketing presented using proprietary noise level plots from loudspeaker manufacturers presented new obstacles requiring further study. DLR Group partnered with Acentech to complete additional site noise measurements and CadnaA environmental noise computer modeling of the site with the potential future improvements. Results of the latest modeling are presented, as well as comparison and discussion of the different modeling programs outputs.

TUESDAY MORNING, 3 DECEMBER 2019 WILDER, 7:45 A.M. TO 11:55 A.M.

Session 2aPA

Physical Acoustics, Structural Acoustics and Vibration, and Computational Acoustics: Design of Acoustics Metamaterials: Optimization and Machine Learning I

Feruza Amirkulova, Chair
Mechanical Engineering, San Jose State University, 1 Washington Sq, San Jose, California 95192

Chair's Introduction—7:45

Invited Papers

7:50

2aPA1. Inverse design of acoustic devices: From flat lenses without negative refraction to acoustic cloaks based on scattering cancellation. Jose Sanchez-Dehesa (Dept. Electron. Eng., Universitat Politècnica de València, Camino de vera s.n., Valencia 46022, Spain, jsdehesa@upv.es), Lorenzo Sanchis (Dept. Electron. Eng., Universitat Politècnica de València, Valencia, Spain), Peter R. Andersen, and Vicente Cutanda Heniquez (Ctr. for Acoustic-Mech. Micro Systems, Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

Inverse design is currently applied to obtain unusual acoustical devices based on ordered and non-ordered scatterers. Recently, flat and lenticular acoustic lenses, de-multiplexors, directional sound sources and acoustic cloaks have been designed using a variety of optimization methods like genetic algorithms, simulated annealing and shape or topology optimizations among others. For example, feasible one-directional cloaks were first designed in two- and three-dimensions using high symmetry objects like cylinders [*Appl. Phys. Lett.* **99**, 074102 (2011)] and toroidal scatterers [*Phys. Rev. Lett.* **110**, 124301 (2013)], respectively. Recently, an extraordinary simple one-directional acoustic cloak has been reported using a technique that combines the method of fundamental solutions with arbitrary shape scatterers [*Sci. Rep.* **8**, 12924 (2018)]. Here, we report a further improvement of the method by combining the Boundary Element Method (BEM) with shape optimization to obtain quasi-omnidirectional cloaks in two-dimensions. The shapes of the scatterers are optimized for the cloaking of the whole setup with waves impinging on the stealth object (a cylinder) from many different directions. An important feature of the method is the possibility of including visco-thermal acoustic losses in the optimization process [*J. Sound Vib.* **447**, 120–136 (2019)].

8:10

2aPA2. Inverse design and deep learning for phononic crystals. Ankit Srivastava (MMAE, IIT Chicago, 10 West 32nd St., John T. Rettaliata Eng. Ctr., Rm. 243, Chicago, IL 60616, asriva13@iit.edu)

In this talk, I will present some results pertaining to the inverse design of phononic crystals. First, I will consider the design of 3-D phononic crystals exhibiting large omnidirectional bandgaps within the context of topology optimization using the SIMP method. The problem will be presented in the traditional forward-inverse paradigm where the forward solver is a mixed-variational scheme implemented on multiple graphical processing units. The inverse solver is a large scale adjoint based optimization framework with highly efficient vectorized operations for sensitivity calculations. In the second part of the talk, I will consider a deep learning based surrogate model for the forward problem. The neural network architecture is based on convolutional neural networks (CNN) and is trained to predict the phononic eigenvalues of unseen unit cell configurations. I show that a CNN based model easily outperforms traditional neural network architectures such as the multi layer perceptron (MLP) on both efficiency of learning and generalization capabilities. Strong and highly efficient surrogate models such as CNN coupled with topology optimization provide a an important way forward for the inverse design of phononic crystals and metamaterials.

8:30

2aPA3. Optimization of quasi-periodically ribbed structures to improve stealth and discreteness of underwater vehicles. Bertrand Dubus (CNRS, Centrale Lille, ISEN, Univ. Lille, Univ. Valenciennes, UMR 8520 IEMN, 41 boulevard Vauban, Lille Cedex 59046, France, bertrand.dubus@isen.fr), Charles Croënne, Samuel Deleplanque, and Anne-Christine Hladky-Hennion (CNRS, Centrale Lille, ISEN, Univ. Lille, Univ. Valenciennes, UMR 8520 IEMN, Lille, France)

Acoustic stealth and discreteness of underwater vehicles are important issues in underwater acoustics. These vehicles are constituted of shells which are periodically reinforced by ribs. Acoustic fields radiated or scattered by such structures are strongly affected by phenomena related to this periodicity such as Bragg diffraction and Bloch-Floquet waves. The introduction of a slight modification of each rib is considered here to reduce this signature for radiation problem, scattering problem or both radiation and scattering problems. Geometrical and material parameter variations used to obtain these new solutions may also be constrained by technological limitations. In a classical process, the design would be realized by experts using tools and methodologies that have been developed over the years. Another approach of the problem consists in using optimization algorithms (genetic, greedy, etc.) which have to be coupled to simulation models of the acoustic radiation and scattering problems. This approach is considered here in the case of a plate periodically reinforced by ribs. Simulation results are reported and the impact of the choice of the objective function, the parameter to be varied, the simulation model and the optimization algorithm on the final performance are also discussed.

8:50

2aPA4. Multi-holographic metamaterials: The concept, artificially intelligent designs, and new applications in acoustics. Yaroslav Urzhumov (Invention Sci. Fund, Intellectual Ventures, 3150 139th Ave. SE, Invention Sci. Fund, Bellevue, WA 98005, yaroslav.urzhumov@duke.edu)

Acoustic holographic metamaterials present a powerful new platform for manipulating acoustic wave fields in fluids and solids. Furthering their development, we introduce a new concept of multi-holographic metamaterials (MHM)—complex media storing multiple volumetric holograms. Such media lack any distributed electromechanical components, yet offer dynamic functionality using only a few electronic components. This novel engineering paradigm shifts device fabrication cost from hardware to computation, presenting the designer with an NP-hard problem of resolving the complex relationship between structure and physical properties. Focused on airborne ultrasonic metamaterials, we treat them as a free-form composite of a fluid and solid(s). This unlocks a huge number of structural degrees of freedom, compared to the standard treatment of metamaterials as periodic arrays of geometrically similar unit cells. While analytical methods for finding all optima of an analytical function are scarce and largely impractical (Groebner basis computation being a notable example), we turn to machine learning methodologies to design MHMs efficiently. We show that unsupervised machine learning, a basic form of Artificial Intelligence, can completely replace human input into the design of MHMs. Experimental demonstrations include airborne ultrasonic metamaterials for coherent beamforming, with applications ranging from directive speakers to contactless manipulation of objects.

9:10

2aPA5. Inverse design method for acoustic metamaterials. He Gao (Mech. Eng., Hong Kong Polytechnic Univ., Hung Hom, Hong Kong) and Jie Zhu (Mech. Eng., Hong Kong Polytechnic Univ., FG603, Hong Kong Polytechnic University, Kowloon 00000, Hong Kong, jiezhu@polyu.edu.hk)

When designing acoustic metamaterials for complex environments, it is very challenging to derive the acoustic field distribution analytically and obtain the desired devices with the conventional method. Here, we propose to optimize some acoustic devices with inversely design method, which can actively find the optimized structure parameters with prescribed property. This inverse design method avoids the complex analytical calculation and can obtain some performances that are not attainable with the conventional method. This inverse design method takes a significant step towards realizing practical devices and would open up new possibilities for manipulating sound wavefront with optimized performances.

9:30

2aPA6. Evolutionary algorithm-based design of acoustic metamaterials for thermal insulation. S. Hales Swift (Energy Systems Div., Argonne National Lab., N221 ESC, Provo, UT 84602, hales.swift@gmail.com), Ralph T. Muehleisen, Raghuveer Chimata (Energy Systems Div., Argonne National Lab., Lemont, IL), Matthew J. Cherukara, Troy D. Loeffler, and Subramanian K. Sankaranarayanan (NanoSci. and Technol. Div., Argonne National Lab., Lemont, IL)

Heat transfer at the nanoscale is an acoustic problem. In nonmetallic solids, heat is primarily carried by lattice vibrations (phonons), and its transfer can similarly be impeded by methods for manipulating vibrations familiar to acousticians. Metamaterials have seen increasing use in altering and controlling vibration. Metamaterial concepts, including periodic structuring of materials and introduction of resonant inclusions, have been used to control vibration on the macroscale, and can also affect heat transfer when implemented at nanoscale. Locally resonant materials appear to be able to address the frequency range responsible for heat transfer most easily. Control of heat transfer is of particular importance because heating and cooling buildings are among the most extensive human uses of energy, thus presenting a significant opportunity for societal benefit through increased efficiency. Even modest improvements in building insulation will result in significant energy and cost savings. In pursuit of a superior and cost-effective insulating material for building applications, an optimization of the nanostructure of a polymer metamaterial panel to reduce heat transfer is carried out using simulation in COMSOL and a genetic algorithm. Results of this research are presented here. [Funded by USDOE Office of Energy Efficiency and Renewable Energy (EERE)—Office of Building Technologies.]

Contributed Paper

9:50

2aPA7. An integrated framework for the design of aeroacoustic metamaterials. Umberto Iemma (Eng., Roma Tre Univ., via della Vasca Navale 79, Roma 00146, Italy, umberto.iemma@uniroma3.it) and Peter Göransson (Aeronautical & Vehicle Eng., KTH, Stockholm, Sweden)

This paper deals with the development of an integrated framework for the development of acoustic metamaterials tailored to aeroacoustic applications. Indeed, the contributions available in the literature targeted are still limited (although constantly increasing) with respect to the gigantic amount of works on standard acoustic metamaterials. The research is driven by the urgent need to substantially reduce the noise produced by the civil aviation to guarantee the sustainable evolution of the system in a constantly growing market. The target reductions required to mitigate the impact on population will be achieved only through the introduction of breakthrough technologies. The present research aims at the complete disclosure of the acoustic metamaterials potential to contribute to this target. The approach starts from the formulation of the fundamental equations governing the propagation of an acoustic disturbance within a generic metacontinuum operating in the presence of a background flow. The model is coupled with an original homogenization scheme, to identify, through numerical optimization, the basic properties of a periodic structure matching the target metacontinuum response. Preliminary numerical results are obtained using dedicated finite and boundary element methods.

10:05–10:20 Break

Invited Papers

10:20

2aPA8. Optimization on metasurface-enabled sound absorbing panels. Yun Jing (North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695, yjing2@ncsu.edu) and Jun Ji (North Carolina State Univ., Raleigh, NC)

An optimization scheme based on the genetic algorithm is proposed to achieve a desired sound absorption spectrum using coupling of multiple resonators at a minimum thickness. The proposed optimization strategy considers the area of supercell, the number of unit cells in a supercell and the shape of each unit cell. Besides the optimization based on the mechanism of coupled resonances, topology optimization with thermoviscous losses being taken into account is further proposed to explore the absorption performance of metasurface-enabled sound absorbing panels. A simplified set of equations and a relatively simple interpolation for the distribution of design variables make the topology optimization an viable option.

10:40

2aPA9. Practical realisation of an active acoustic metamaterial building block. 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 (AMMs) have been demonstrated as an alternative approach to achieving high levels of noise control using an array of subwavelength unit cells, exhibiting behaviour not seen in conventional materials. Specifically, active AMMs offer the potential for greater levels of broadband wave manipulation, tunability and adaptability. However, determining the control source strengths that achieve broadband negative effective material properties is not straightforward. This study presents a practical method of designing an active system that directly minimises the effective material properties. The source strengths required for both single monopole and dipole control sources to minimise the effective bulk modulus and the effective density respectively have been calculated via an optimisation procedure in the frequency domain. A finite impulse response (FIR) filter has then been designed in each case to match the optimised frequency responses and enable real-time implementation. The performance of the designed FIR filters has been tested by implementation using the two different control sources in a one-dimensional duct and the ability of the proposed active AMM building

blocks to achieve broadband negative effective material properties is tested. Interestingly, by combining the two optimised control sources, double negativity can be achieved, offering the potential for negative refraction.

11:00

2aPA10. Convolutional neural network driven design optimization of acoustic metamaterial microstructures. Corbin Robeck (Appl. Sci., Thornton Tomasetti, 2000 L St. NW Ste. 600, Washington, DC 20036, CRobeck@ThorntonTomasetti.com), Jeffrey Cipolla (Appl. Sci., Thornton Tomasetti, Washington, DC), and Alex Kelly (Appl. Sci., Thornton Tomasetti, New York, NY)

The design of broadband mechanical metamaterials in the context of acoustics can be driven by the invariance of the wave equation under a set of special coordinate transformations called transformation acoustics. A program of homogenization to design the metamaterial structure can be derived from these transformation functions subject to geometric constraints from the metamaterial's intended application. The limits of homogenization in a nonperiodic, nonlinear environment however must be accounted for and corrected. This can be done by means of nonlinear manifold interpolation methods, the feedback into which is driven by the scattered wave field in the ambient medium. The feedback is minimized using statistical and machine learning techniques, dictating the final metamaterial structure. The performance of this algorithmic method is compared against a "brute force" optimization approach driven by a convolutional neural network with no transformation of the underlying wave equation.

11:20

2aPA11. Adjoint-based, large-scale optimization of metamaterials. Gregory Bunting (Computational Solid Mech. & Structural Dynam., Sandia National Labs., 709 Palomas Dr. NE, Albuquerque, NM 87108, gbuntin@sandia.gov) and Timothy F. Walsh (Computational Solid Mech. & Structural Dynam., Sandia National Labs., Albuquerque, NM)

Harsh shock and vibration environments are commonly encountered in engineering applications involving dynamic loading. Acoustic/elastic metamaterials are showing significant potential as candidates for controlling wave propagation and isolating sensitive structural components. However, these materials have complex microstructures that must be properly designed to achieve their desired properties. Since each resonator could potentially have its own stiffness/mass properties, the design parameter space for a three-dimensional geometry quickly becomes prohibitively large for global search-based optimization strategies. This brings in the need for adjoint-based optimization. In this talk we will present strategies for adjoint-based, PDE-constrained design optimization of locally resonant elastic/acoustic metamaterials. We will present a variety of resonator geometries that can be easily optimized for wave control applications, along with fabrication details involving multi-material additive manufacturing. A variety of objective functions (time-domain, frequency-domain, and mode shape-domain) will be compared for their effectiveness in designing mechanical filters. [Sandia National Laboratories is a multimission laboratory managed and operated by National Technology and Engineering Solutions of Sandia, LLC., a wholly owned subsidiary of Honeywell International, Inc., for the U.S. Department of Energy's National Nuclear Security Administration under Contract No. DE-NA-0003525.]

Contributed Paper

11:40

2aPA12. Broadband acoustic metamaterial design using gradient-based optimization. Lauren Fahey (Mech. Eng., San Jose State Univ., One Washington Square, San Jose, CA 95192, lauren.fahey@sjsu.edu), Feruza Amirkulova (Mech. Eng., San Jose State Univ., Springfield, MA), and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

We present a semi-direct method for broadband metamaterial design using gradient-based optimization by means of multiple scattering theory. As application of the method we design acoustic metamaterial based on a gradient-based minimization of total scattering cross section (TSCS) from a set of cylinders by incrementally repositioning them so that they eventually act as an effective cloaking device. We obtained an analytical formula for the gradients of the TSCS with respect to positions of a set of cylinders in closed form rather than relying on finite differences. We defined the broadband gradient vectors with respect to positions which can be found in terms of the individual single frequency gradients. The root mean square of a set of TSCS from a set of cylindrical obstacles is minimized over a range of frequencies. As another application we model broadband wide-angle acoustic lens by maximizing the absolute pressure at a focal point and analytically evaluating its derivative with respect to the cylinder positions. We optimize the cost function for a given set of wavenumbers and incident angles, while supplying the gradients. We illustrate how the analytical form of gradients of the objective function enhances the modeling and accuracy of the optimized solution.

Session 2aPP

Psychological and Physiological Acoustics: Current Topics in Physiological and Psychological Acoustics (Poster Session)

Tian Zhao, Chair
Institute for Learning and Brain Sciences, University of Washington, Box 367988, Seattle, Washington 98028

All posters will be on display from 9:00 a.m. to 11:30 a.m. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10:15 a.m. and authors of even-numbered papers will be at their posters from 10:15 a.m. to 11:30 a.m.

Contributed Papers

2aPP1. Auditory brainstem responses to successive sounds: Effects of gap duration and depth. Fan-Yin Cheng (Commun. Sci. and Disord., Univ. of Texas at Austin, 11215 Res. Blvd. Apt. 2037, Austin, TX 78759, fanyin.cheng@utexas.edu), Won So, and Craig A. Champlin (Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX)

Temporal acuity is the ability to differentiate between successive sounds based on temporal fluctuations in the waveform envelope. Psychophysically, human listeners can detect a gap as short as 2.5 ms between consecutive segments to encode the acoustical messages. The background noise diminishes the ability to follow fast variation between segments. In this study, we determined whether a physiological correlate of temporal acuity is also affected by the presence of noise. We recorded the auditory brainstem response (ABR) from human listeners using a harmonic complex followed by tone burst with the latter serving as the evoking stimulus. The duration and the depth of the silent gap between the harmonic complex and tone burst were manipulated. The latency of the ABR increased significantly as gap duration increased and gap depth decreased. No significant changes in amplitude were observed. These findings suggest that changing gap duration and depth affect the auditory system's ability to encode successive sounds.

9:00

2aPP2. Neural tracking of musical rhythm reflected in the autonomic nervous system. Tian Zhao (Inst. for Learning and Brain Sci., Univ. of Washington, Box 367988, Seattle, WA 98195, zhaotc@uw.edu) and Patricia K. Kuhl (Inst. for Learning and Brain Sci., Univ. of Washington, Seattle, WA)

Humans start responding to music even before birth, and one particular measure, heart rate variability (HRV), reflecting the parasympathetic nervous system (PNS) function, has been shown to respond to a music stimulus and to music therapy in the NICU. However, whether and how HRV is related to musical rhythm processing has not been examined. A group of adults ($N = 15$) passively listened to duple and triple rhythmic patterns while we measured their neural tracking of the rhythmic patterns and PNS function simultaneously using MEG and ECG. Results showed that adults' neural tracking of beat level rhythm is 2.5 times stronger than their tracking of meter level rhythm, regardless of the energy of the beat and meter in the stimulus. Moreover, this ratio is significantly related to HRV across individuals. That is, the stronger an individual's brain tracks the meter level rhythm, the higher the HRV, indicating a higher level PNS function (i.e., lower stress). This relation is not affected by an individual's experience with music training. Data collection in infants (7-month and 11-month) is currently ongoing and the results will be discussed in relation to the adult findings.

2aPP3. Physiological measurements of acoustic trauma following blast-induced traumatic brain injury (TBI) in laboratory mice. Kali Burke (Psych., Univ. at Buffalo, SUNY, 246 Park Hall, Buffalo, NY 14260, kaliburk@buffalo.edu), Senthilvelan Manohar, Richard J. Salvi (Dept. of Communicative Disord. and Sci., Univ. at Buffalo, SUNY, Buffalo, NY), and Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, Buffalo, NY)

Exposure to a high intensity blast from the detonation of improvised explosive devices (IEDs) can lead to a traumatic brain injury (TBI) and concurrent trauma to major auditory structures. Research on whether this trauma is permanent or temporary has been inconclusive due to heterogeneity of injury acquisition. A blast wave system that generates consistent and controllable blast waves has been developed, allowing for investigation into the co-morbidity of auditory damage and TBI. In this study, the effects of blast exposure on auditory brainstem responses (ABRs) and distortion product otoacoustic emissions (DPOAEs) in male and female mice were measured. ABRs and DPOAEs were collected prior to exposure to a blast to represent the baseline auditory capacity of each individual subject. Mice were tested at 3, 30, and 90 days after the exposure to capture initial trauma and the degree of recovery. A shift in ABRs and DPOAEs thresholds was observed in post-exposure mice, indicating trauma to central and peripheral auditory systems. Following the 90 day tests, brains and cochleae were collected for further analysis. [Work supported by R01DC016641.]

2aPP4. Multilevel objective measures of interaural phase difference detection. Spencer Smith (Commun. Sci. and Disord., The Univ. of Texas at Austin, 39 Woodstone Square, Austin, TX 78703, spencer.smith@austin.utexas.edu) and Won So (Commun. Sci. and Disord., The Univ. of Texas at Austin, Austin, TX)

Sensitivity to interaural phase difference (IPD) cues is critical for localizing sound and hearing in background noise. Recent work from multiple laboratories has explored electrophysiologic methods for assaying IPD sensitivity with the aim of developing an objective measure of binaural hearing. Such a measure would be useful in a variety of clinical and research applications such as understanding the effects of aging on binaural hearing; fine-tuning hearing aids/cochlear implants to maximize binaural benefit; and assessing neural damage after a traumatic brain injury. Here, we present data from an ongoing study evaluating a novel "multi-level" (brainstem, midbrain, and cortex) simultaneous measure of IPD acuity. Amplitude modulated (20, 40, and 80 AM) carrier tones (125, 250, and 500 Hz) with IPDs embedded in them (7 deg–180 deg) were used to generate auditory evoked potentials in young, normal hearing listeners. Neural responses corresponding to the carrier tones (frequency following responses), amplitude modulation envelopes (envelope following responses), and cortical change detection (acoustic change complex) were measured simultaneously to provide a multi-level snapshot of neural IPD processing. Parametric data

presented here demonstrate feasibility and limitations of this approach for measuring IPD encoding at multiple levels of the auditory system.

2aPP5. Effects of interaural phase on the frequency following response.
Won So (Commun. Sci. and Disorder, Univ. of Texas at Austin, 1628 W 6th St Apt BApt C107, Austin, TX 78703, wonso1983@gmail.com), Fan-Yin Cheng, and Craig A. Champlin (Commun. Sci. and Disorder, Univ. of Texas at Austin, Austin, TX)

The frequency following response (FFR) is generated by periodic auditory stimuli and represents the summed phase-locked activity from the brain. The aim was to examine this neural activity in response to several interaural phase configurations of binaural stimulation. The FFR was recorded from 10 young adults with normal hearing. The stimulus was a 400-Hz tone burst presented at 70 dB SPL. The starting phase of the tone presented to the right ear was always 0 deg while the phase of the tone presented to the left ear varied in 45-deg increments from 0 to 180 deg. The FFR amplitude measured with an interaural phase of 0 deg was not significantly different from amplitudes measured at 45 and 90 deg; however, FFR amplitudes acquired at 135 and 180 deg were significantly lower than the 0-degree condition. The dominant frequency component of the FFR was 400 Hz in all interaural phase conditions except 180 deg. The implications of neural measures of binaural interaction will be discussed.

2aPP6. Pupil-associated states modulate excitability but not stimulus selectivity in primary auditory cortex.
Zachary P. Schwartz (Neurosci. Graduate Program, Oregon Health & Sci. Univ., 3181 SW Sam Jackson Park Rd. MC L335A, Portland, OR 97239, schwarza@ohsu.edu), Brad N. Buran, and Stephen V. David (Oregon Hearing Res. Ctr., Oregon Health and Sci. Univ., Portland, OR)

Recent research indicates that luminance-independent fluctuations in pupil size predict variability in spontaneous and evoked activity of single neurons in auditory and visual cortex. These findings suggest that pupil is an indicator of large-scale changes in arousal state that affect sensory processing. It is not known whether pupil-related state also influences the selectivity of auditory neurons. We recorded pupil size and single-unit spiking activity in the primary auditory cortex (A1) of ferrets during presentation of vocalizations and tone stimuli. Neurons showed a systematic increase in both spontaneous and sound-evoked activity when pupil was large, as well as a decrease in trial-to-trial variability. Relationships between pupil size and firing rate were non-monotonic in some cells. In most neurons, several measurements of tuning, including acoustic threshold, spectral bandwidth, and best frequency, remained stable across large changes in pupil size. Across the population, however, there was a small but significant decrease in acoustic threshold when pupil was dilated. In some recordings, we observed rapid, saccade-like eye movements during sustained pupil constriction, which may indicate episodes of rapid eye movement sleep. Including the presence of this state as a separate variable in a regression model of neural variability accounted for some, but not all, of the variability associated with changes in pupil size.

2aPP7. Ripple depth and density thresholds: Estimate dependence on the measurement paradigm.
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 signals are a convenient test to measure spectro-temporal resolution of auditory stimuli. There is no commonly adopted paradigm for measurements of rippled-spectrum resolution. Test stimuli may be either rippled-spectrum signals with ripple phase changes during the signal or constant-phase rippled-spectrum signals. Reference stimuli may have either rippled or non-rippled spectra. In the present study, a direct comparison revealed that for rippled reference signals, the ripple depth thresholds were as low as 0.1 at low ripple densities; for a maximum ripple depth of 1.0, the ripple density resolution was 8 to 9 ripple/oct. For nonrippled reference stimuli, the ripple density resolution was as high as 25–30 ripple/oct; however, ripple depth thresholds were not less than 0.5 at ripple densities above 7 ripples/oct. It is hypothesized that the differences between resolution estimates obtained with different reference stimuli appear due to exploiting different cues for discrimination between test and reference stimuli, specifically, the cues provided by the excitation-pattern and temporal-processing mechanisms of frequency analysis. Discrimination between rippled test and rippled reference signals agrees with an excitation-pattern model, whereas discrimination between rippled test and non-rippled reference stimuli agrees with a temporal-processing model. [Work supported by Russian Science Foundation, Grant No. 16-15-10046.]

2aPP8. Temporal loudness weights in background noise: Data and models.
Daniel Oberfeld (Experimental Psych., Johannes Gutenberg-Universität Mainz, Wallstrasse 3, Mainz 55122, Germany, oberfeld@uni-mainz.de), Alexander Fischenich (Experimental Psych., Johannes Gutenberg-Universität Mainz, Mainz, Germany), Jesko L. Verhey, and Jan Hots (Experimental Audiol., Otto von Guericke Univ. Magdeburg, Magdeburg, Germany)

Previous studies consistently showed that human listeners primarily consider the beginning of a time-varying sound when judging its overall loudness, and place less weight on subsequent temporal portions. However, all experiments studying this primacy effect in temporal loudness weights presented the target sound in quiet. Here, we compared temporal weights when the target sound was either presented in quiet or in a continuous background noise, and for a variation in the level of the target sound across a range of 60 dB. The target sound was a time-varying narrowband noise, the background noise was a continuous bandpass-filtered noise. In all conditions, we observed the expected primacy effect, well described by an exponential decay function using parameters based on previous studies. The patterns of temporal weights were very similar in conditions with and without background noise, and largely independent of the variation in target level. Simulations using a dynamical model for the partial loudness of time-varying sounds in background noise showed that the model does not predict the observed temporal loudness weights. Strongly depending on the loudness statistic extracted from the loudness model, the predicted loudness weights were either uniform, or showed an inverse U-shaped pattern with higher weights predicted for temporal portions in the middle of the sound compared to temporal portions near onset or offset.

2aPP9. Altered auditory sampling in infants at risk for dyslexia across the sensitive period of native phoneme learning.
Maria Mittag (Inst. for Learning & Brain Sci., Univ. of Washington, Portage Bay Bldg., Rm. 356, Seattle, WA 98195-7988, mmittag@uw.edu), Eric Larson, Samu Taulu, Maggie Clarke, and Patricia Kuhl (Inst. for Learning & Brain Sci., Univ. of Washington, Seattle, WA)

Recent research in dyslexia postulates that imprecise representations of phonemes are linked to basic auditory deficits in temporal sampling. Evidence comes from adult readers with dyslexia demonstrating deficits in auditory sampling at several rates of the speech signal including the phoneme rate (>40 Hz). It remains unknown whether such deficits are present in infancy, especially during the "sensitive period" for native phoneme learning (6–12 months). We examined auditory sampling in infants at risk for dyslexia across the period of native phoneme learning. Using magnetoencephalography, we recorded auditory steady-state responses (ASSRs) to white noise that was amplitude-modulated at frequency rates relevant to speech processing in 6- and 12-month-old at-risk and typically developing (TD) infants. Our results show that the two groups differed in hemisphere involvement of ASSRs at the phoneme rate (50–80 Hz): TD infants showed a shift from right-to-left hemisphere processing from 6 to 12 months, whereas at-risk infants showed stronger right-hemisphere processing at 12 than 6 months. This measure predicted words understood at 13 months. Risk for dyslexia is associated with auditory sampling deficits throughout the sensitive period. These deficits may affect future language acquisition.

2aPP10. Modeling the spatial variations of the intracochlear fluid pressure based on *in vivo* mechanical measurements. Michael Rouleau (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332, mike.rouleau@gatech.edu) and Julien Meaud (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

The mammalian cochlea is a complex system consisting of two fluid-filled ducts separated by a structure known as the organ of Corti. Fluid pressure in the cochlear ducts interact with the organ of Corti to stimulate the mechanoelectrical receptors responsible for hearing. Because, until recently, *in vivo* dynamic measurements were limited to the basilar membrane, most cochlear models consider only basilar membrane—fluid interactions. Recent measurements of tectorial membrane and reticular lamina motions using optical coherence tomography motivate us to develop cochlear models that takes into account separately coupling of the fluid with all these components of the organ of Corti. In this work, finite element models are used to simulate an apical slice of a mouse cochlea. Using experimentally measured motion of the stapes, reticular lamina, basilar membrane, and tectorial membrane [Lee *et al.*, *J. Neurosci.* (2016)] as input, spatial variations of pressure in the cochlea are explored. This study investigates fluid dynamics in the cochlear ducts and explores the relationship between the motion of the organ of Corti and spatial pressure variations while providing a basis for future computational models to consider cochlear fluid dynamics in the entire cochlea.

2aPP11. Basic auditory processing deficits in infants at risk for dyslexia during the sensitive period predict future language. Maria Mittag (Inst. for Learning & Brain Sci., Univ. of Washington, Portage Bay Bldg., Rm. 356, Seattle, WA 98195-7988, mmittag@uw.edu), Samu Taulu, Eric Larson, Maggie Clarke, and Patricia K. Kuhl (Inst. for Learning & Brain Sci., Univ. of Washington, Seattle, WA)

It is widely agreed that poor reading and spelling performances in individuals with developmental dyslexia underlie problems with processing phonetic units. According to recent child and adult data, such processing deficits likely are the consequence of compromised low-level auditory processing. A possible causal relationship can be explored by testing infants based on their familial risk of dyslexia. Using magnetoencephalography, the study at hand investigates low-level auditory processing in infants at risk for dyslexia across 6 and 12 months of age—a sensitive period in early language development when the auditory system specializes in native phoneme perception. Results showed smaller and shorter neural responses to simple sounds in at-risk infants at 6 than at 12 months, a pattern that was reversed in age-matched typically developing infants. This interaction was significant both when fitting equivalent current dipoles and when using distributed source modeling and localized to left temporal and left frontal brain regions, indicating its potential impact on early language learning. In addition, atypical auditory responses in at-risk infants consistently predicted later language skills, such as syntactic processing and word production, thus identifying a possible biomarker of dyslexia in the infant brain.

2aPP12. Serial recall of spectro-temporal patterns under informational masking. Christopher Conroy (Speech, Lang. & Hearing Sci. and Hearing Res. Ctr., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, cwconroy@bu.edu), Elin Roverud, and Gerald Kidd (Speech, Lang. & Hearing Sci. and Hearing Res. Ctr., Boston Univ., Boston, MA)

This study investigated the memory *costs* associated with informational masking (IM) and the memory *benefits* associated with its reduction. A serial recall task was used wherein the stimuli to be recalled were random sequences of six possible learned spectro-temporal patterns masked by other random sequences of spectro-temporal patterns. Sequence lengths of 1, 3, and 5 were tested. In all conditions, a target sequence and a masker sequence were assigned to mutually exclusive frequency bands. In the high-IM reference condition, the frequency separation of the two bands was chosen so as to be just discriminable and the onsets of corresponding target and masker patterns roughly overlapped in time. In the comparison conditions, IM was reduced systematically by increasing the frequency separation of the two bands and staggering the onsets of corresponding patterns. When IM was high, recall was poor and memory costs were large. As IM was reduced, recall improved, revealing memory benefits typically not discernable in standard detection, discrimination, or recognition-based studies of IM. Large individual differences were found and a number of observers were unable to perform the task at the pre-established criterion level even after training in unmasked conditions. The interaction between IM and memory will be discussed.

2aPP13. Polyphonic pitch perception in cochlear implant users. Andres Camarena (Otolaryngol., Univ. of Southern California, 1640 Marengo St., Los Angeles, CA 90089, andresc@usc.edu) and Raymond L. Goldsworthy (Otolaryngol., Univ. of Southern California, Los Angeles, CA)

Cochlear implant (CI) devices take advantage of the tonotopic organization of the auditory nerve to provide moderate speech recognition. While commercial place-pitch strategies are sufficient for speech comprehension in quiet conditions, CI users struggle to detect multiple, simultaneous sources of pitch. In a polyphonic pitch-trajectory task, two streams (low and high pitch) are presented simultaneously. Participants perform a 1-interval-4-alternative forced-choice procedure to identify which stream changed in pitch and whether it went higher or lower in pitch (low-lower/low-higher/high-lower/high-higher). The low and high streams are high-rate pulsetrains modulated at 110 Hz and 220 Hz, respectively. The stimulus is 1 s in duration and the target stream contains an ascending or descending modulation-frequency sweep over the last 200 ms. The low stream will be presented to the most apical electrode and the high stream presented to electrode separations of 0, 2, 4, and 8. Preliminary results indicate that CI users perform better on the electrode psychophysical tasks than on acoustic variations. Additionally, performance is improved with sharper temporal envelopes. These results suggest that CI users are able to perceive polyphonic pitch and that resolution is limited by how temporal information is conveyed by clinical devices.

2aPP14. Establishing an audibility sensitivity rule for low-frequency tone complexes in noise. Menachem Rafaelof (National Inst. of Aerosp., M.S. 463, 100 Exploration Way, Hampton, VA 23681-2199, menachem.rafaelof@nasa.gov), Andrew Christian, Kevin P. Shepherd, Stephen A. Rizzi (NASA Langley Res. Ctr., Hampton, VA), and James H. Stephenson (US Army Aviation Development Directorate, Hampton, VA)

Predicting the audibility of a sound over ambient noise is an important step to assess its impact on observers. One anticipated need for such predictions is the desire to operate aerial vehicles in urban surroundings close to the public. The increased audibility of a complex set of low-frequency tones due to the recruitment of a number of independent auditory channels has been investigated in this work. Three experiments were carried out to establish a sensitivity rule for these signals. During the main experiment, the masked threshold for signals consisting of one, two, and three tone components (55, 120, and 200 Hz) were measured. As expected, the thresholds identified during this experiment point to an increased sensitivity relative to a single-tone case. However, these increases were greater than similar findings by other researchers at higher frequencies. Two follow-up experiments examined the effect of larger frequency separation between tones and their intermodulation. The findings of these experiments attribute part of the audibility improvement to biased estimation of the thresholds of multitone sounds due to an error in the estimation of thresholds for single tones. Additional details may be found in a forthcoming NASA Technical Memorandum entitled "Audibility of Multiple, Low-Frequency Tonal Signals in Noise."

2aPP15. Ripple spectrum resolution by prelingual and postlingual cochlear implant users. Dmitry Nechaev (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, dm.nechaev@yandex.ru), Marina Goykhburg (National Res. Ctr. for Audiol. and Hearing Rehabilitation, Moscow, Russian Federation), Alexander Supin (Inst. of Ecology and Evolution, Moscow, Russian Federation), Vigen Bakhshinyan, and George Tavartkiladze (National Res. Ctr. for Audiol. and Hearing Rehabilitation, Moscow, Russian Federation)

The correlation between ripple spectrum resolution and speech recognition was investigated for prelingual and postlingual cochlear implant users. The test signal had a band limited rippled spectrum (spectrum with

alternated maxima and minima of spectral power) with equivalent rectangular bandwidth of 1 oct. The central frequency of the spectrum was 1, 2, or 4 kHz. The principle of rippled-spectrum test was to find the maximum ripple density (ripples/octave) at which listeners could detect the phase reversion of the spectrum. A balanced test set of Russian worlds by Greenberg-Zinder was used for estimation of speech recognition. The average ripple discrimination thresholds were approximately 2 ripples/octave in both pre- and postlingual patients. The thresholds did not depend from central frequency. The best correlation between speech recognition and rippled spectrum resolution was found in prelingual patients with perimodiolar electrode array compared with straight array (r = 0.85). In postlingual patients, correlation was markedly lower (r = 0.44 and r = 0.13 for rippled-spectrum tests centered at 1 or 2 kHz, respectively). [Work supported by the Russian Science Foundation, Grant No. 16-15-10046.]

2aPP16. Effects of steady background noise on binaural pitch fusion. Sabrina Lee (Speech, Lang., and Hearing Sci., Univ. of Florida, 1600 SW Archer Rd., Rm. D2-77, Gainesville, FL 32610, sabrina.lee@ufl.edu), Yi Yuan, and Yonghee Oh (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

Binaural pitch fusion is the fusion of dichotic stimuli across the two ears. The current study was designed to investigate how steady background noise can influence binaural fusion. The binaural fusion ranges, the frequency ranges over which binaural fusion occurred, were measured with various levels (30—70 dB/ERB$_N$) of the threshold-equalizing noise (Moore *et al.*, 2004) and compared with those measured in quiet. Two interleaved adaptive procedures were used to estimate the fusion ranges at the reference frequencies of 2 and 3 kHz: Single-Interval Adjustment Matrix (SIAM; Kaernbach, 1990) and Maximum-Likelihood (ML; Green, 1992). The results show that the TEN decreases the fusion ranges to about 1.5 to 2 times narrower than those in the quiet condition. Even soft noise levels (i.e., the 30 dB/ERB$_N$) significantly decreased fusion ranges. For some subjects, the fusion range asymptoted with a soft level and did not increase with increased noise level; for other subjects, the fusion range minimized only at the specific noise level. Both SIAM and ML procedures showed similar performance in fusion range estimation. The findings suggest that the importance of steady background noise, specifically TEN, in binaural fusion may vary across subjects. This may provide potential rehabilitation approaches to reduce broad fusion in hearing impaired listeners.

2aPP17. Changes in sound lateralization ability of elderly individuals when the onset conditions of pure tone of 1 kHz are varied. Kazumoto Morita (Chuo Univ., 1-13-27, Kasuga, Bunkyo-ku, Tokyo 112-8551, Japan, mtkkojiro@gmail.com), Tsukuru Osawa (Fmr. Chuo Univ., Bunkyo-ku, Tokyo, Japan), and Takeshi Toi (Chuo Univ., Bunkyo-ku, Tokyo, Japan)

In our previous experiments, we found that the recognition of onset of the sound with Interaural Time Differences (ITDs) was inferior for the elderly in comparison with the younger participants. In the present experiment, we have conducted a test using 17 elderly and 16 young participants, in which the duration of the cosine-shaped onset of a 1 kHz pure tone that lasted one second was changed to 4 types of 1, 2, 5 and 10 ms. In addition, 9 types of ITDs (0.2, 0.4, 0.6, 0.8 ms leading to the left or right ear besides the same condition to both ears) were set. In the experiment, a reference sound indicating the front was presented followed by a test sound. Participants responded with 3 choices of the direction of the test sound compared with the reference sound: “Left,” “Same,” or “Right.” The following findings were obtained. (1) Elderly participants are more likely than younger participants to poorly lateralize when sound precedes to the right ear first than to the left. (2) For the elderly participants, the longer the cosine-shape of the onset is, the more likely direction is to be erroneously recognized.

2aPP18. Validation of the whisper test modified by distance as a screening test of hearing impairment for young adults. Nyilo Purnami, HMS Wiyadi, Rosa Falerina (Otorhinolaryngology Head and Neck Surgery, Universitas Airlangga-Dr. Soetomo General Academical Hospital, Surabaya, East Java, Indonesia), Ainun Nadiroh (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia), Dhany Arifianto (Dept. of Eng. Phys., Institut Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id), and Puguh S. Nugroho (Otorhinolaryngology Head and Neck Surgery, Universitas Airlangga-Dr. Soetomo General Academical Hospital, Surabaya, Indonesia)

Pure-tone audiometry as a gold standard is difficult to perform in several places in Indonesia due to problems related to access, referral systems, and costs. Therefore, the examiner relies on the whisper test as a screening test. In this study, we evaluated the newly proposed method of whisper test by modifying the distance so that this method quite practical, simple and can be performed in a smaller room and for the large groups. The test was conducted to 618 randomly selected students (young adults). Sensitivity, specificity, positive predictive value (PPV), and negative predictive value (NPV) of the screening test were compared with pure tone audiometry. Based on diagnostic and screening test evaluation, the results showed that the PPV and NPV values are 13.43% and 99.27% respectively. The whisper test modified by distance yielded high sensitivity (69.23%) and high specificity (90.41%) for the screening test. This hearing screening method will be useful in identifying more hearing impairment for a huge population which helps in referral planning.

2aPP19. The multisensory benefit of informative sound in visual task performance. Alexandra L. Bruder (Vanderbilt Univ., 1211 Medical Ctr. Dr., Nashville, TN 32203, alexandra.l.bruder@vanderbilt.edu), Judy Edworthy (Univ. of Plymouth, Plymouth, United Kingdom), Joseph Schlesinger (Vanderbilt Univ. Medical Ctr., Nashville, TN), and Clayton Rothwell (The Ohio State Univ., Columbus, OH)

Auditory alarms are relied upon to provide cues for industries with high-risk, multisensory performance tasks such as health care. In an anechoic chamber, anesthesiology residents (N = 25) were tested in a simulated multi-task setting, including a patient monitoring primary task with alarmed events, and a visual vigilance task. Alarm type was varied between conventional (following the International Electrotechnical Commission Standard 60601-1-8) and a novel auditory icon alarm, which provide additional information about the event causing the alarm. Novel alarm usage led to a 37% increase in vigilance accuracy and 160 ms reduction in response time, implying that the use of auditory icon alarms can provide multisensory benefits. These findings suggest that novel auditory icons help individuals by reducing the cognitive burdens of primary tasks through reducing visual search of the patient monitoring display. Therefore, auditory alarms help to reduce visual demand through offering cues to change; attention can be focused through the recognition of specifically encoded characteristics of audiovisual objects. These findings advocate for reconsideration of alarm type usage in favor of novel alarms, especially in high-stakes environments to potentially improve patient safety and outcomes.

2aPP20. A study on the ASMR effect of reed wind sound. Ik-Soo Ann (Commun. Eng., Soongsil Univ., 369 Sangdo-ro, Dongjak-gu, Seoul 156-743, South Korea, aisbestman@naver.com) and Myungjin Bae (Commun. Eng., Soongsil Univ., Seoul, South Korea)

The ASMR (Autonomous Sensory Meridian Response) method using sound is an acoustic healing method that stimulates the human hearing by using the sound during the five senses, thereby making the mind comfortable. The acoustic characteristics of the ASMR are being used in various ways as a very quiet and calm sound. In this paper, ASMR effect of reed

wind sound is studied. Experimental results show that the sound of reeds in the reed forests that run across each other by the wind shows white noise components evenly distributed over the entire bandwidth of the human audible frequency band. To check the effect of reed wind on people, we tested physical information such as blood pressure and pulse, and conducted interviews and surveys. The results of the physical examination performed while listening to the reed winds showed that healing of the mind and body was comfortable. The result of the MOS test and the interview showed that the sound of the reed winds drove the thought out and the body and mind seemed to relax. Through this study, it was numerically proved that the reed winds produced by the reed forest winds have the effect of comforting and healing people.

2aPP21. Autism? Robert H. Cameron (Eng. Technol., NMSU (retired), 714 Winter Dr., El Paso, TX 79902-2129, rcameron@elp.rr.com)

This poster explores the acoustic space surrounding an eight year old autistic child, wearing a shooters active hearing protection headset. When his mother was worried about his inability to focus on his homework and his frustrated response to speech therapy, I gave her the hearing protection I wear when I shoot. This headset has about a 34db passive acoustic attenuation in each side and in only one side an added amplifier microphone and loudspeaker which bring the level in that side to the same as without the protection. The result was a child who could focus on his homework.

2aPP22. Ambient noise is "the new secondhand smoke." Daniel Fink (The Quiet Coalition, P.O. Box 533, Lincoln, MA 01733, DJFink@thequietcoalition.org)

Ambient noise is "the new secondhand smoke." Like secondhand smoke, excessive ambient noise is not just a nuisance but is a health and public health problem for millions of Americans. Excessive ambient noise causes hearing loss; disrupts sleep, function, and communication; and causes non-auditory health effects including cardiovascular disease and death. The sounds that matter to people are the ones reaching their tympanic membranes, or perhaps the cochlear hair cells or auditory processing cortex. Specific evidence-based noise exposure levels affect human health. 30 dBA ($L_{Aeq(8)}$) disrupts sleep. 45 dB (L_{dn}) disturbs concentration. 55 dB daily average (L_{den}) has non-auditory heath effects, including cardiovascular disease and death, due to stress responses including activation of the autonomic nervous system and neuroendocrine axis. Approximately 60 dBA (L_{Amax}) interferes with speech comprehension for the hearing impaired. 70 dBA daily average ($L_{Aeq(24)}$) causes hearing loss, and 70 dBA (L_{Amax}) interferes with speech comprehension for those with normal hearing. 85 dBA ($L_{Aeq(8)}$) is the occupational recommended exposure level in the United States, and 85 dBA ($L_{Aeq(1)}$) is the recommended exposure limit to prevent hearing loss. Acoustic scientists and engineers must be aware of these specific, evidence-based noise exposure levels and must work to reduce excessive ambient noise.

Session 2aSA

Structural Acoustics and Vibration and Noise: Flow-Induced Vibration and Noise

Kuangcheng Wu, Cochair
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Chair's Introduction—8:45

Invited Papers

8:50

2aSA1. Aeroacoustic source localization and source level estimation during wind tunnel testing of a flat plate with and without a gap. Alexander S. Douglass (Mech. Eng., Univ. of Michigan, Ann Arbor, MI) and David R. Dowling (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109-2133, drd@umich.edu)

Acoustic measurements in wind tunnels are notoriously difficult because of machinery noise from the tunnel, and aeroacoustic noise generated within the tunnel and by the test model; and because the potentially-weak signals of interest may share a common frequency range with these noise sources. Thus, when model changes are made, localizing any new aeroacoustic source(s) and determining their level(s) are challenging tasks. This presentation provides such experimental results for the aeroacoustic source associated with addition of a 6 mm gap in a 0.5-m-by-1.0-m flat plate aligned with the flow direction. When present, the gap was located 0.40-m from the plate's leading edge. The measurements were collected in the Anechoic Flow Facility of the Naval Surface Warfare Center – Carderock Division at nominal air speeds of 19 and 30 m/s using a ~0.6-m-diameter spiral microphone array with 24 elements placed 1.24 m from the plate and separated from the air flow by a thin barrier. Measurements in the 4-to-12 kHz frequency range were processed using conventional and high-resolution beamforming methods with and without noise-reference subtraction. Source location and level estimation for the gap-induced aeroacoustic sound source was successful at nominal signal-to-noise ratios of –20 dB. [Work supported by NAVSEA through the NEEC.]

9:10

2aSA2. Investigation of the unsteady responses of airfoils due to small-scale motion. Matthew R. Catlett (Naval Surface Warfare Ctr. - Carderock Div., 9500 MacArthur Blvd, West Bethesda, MD 20817, matthew.catlett@navy.mil), Jason Anderson (Naval Surface Warfare Ctr. - Carderock Div., West Bethesda, MD), and James Baeder (Univ. of Maryland - College Park, College Park, MD)

Unsteady pressures, forces, and pitching moments generated by foils experiencing vibratory motion in an incompressible, attached flow configuration are studied within this work. Specifically, two-dimensional, unsteady potential flow calculations are performed on Joukowski foils of varying thickness undergoing variable amplitude, small-scale, heaving or pitching motion over a range of reduced frequencies between 0.01 and 100. These calculated results are compared directly to predictions from implementing the Theodorsen model, which treats foils as infinitely thin flat plates that shed a planar sheet of vorticity. The effects of relaxing these seemingly strict assumptions of the Theodorsen model are explored, and focus is placed on results which show deviations from the Theodorsen model. These include altered unsteady responses for finitely thick foils and non-zero mean angle of attack conditions, non-linearity of the wake of shed vorticity, and analysis of the unsteady streamwise force. Further, within the potential flow framework the particular terms which control the unsteady responses are identified.

9:30

2aSA3. Structural vibration of an elastically supported plate due to excitation of a turbulent boundary layer. Nickolas Vlahopoulos (Univ. of Michigan, 2600 Draper Dr., Ann Arbor, MI 48109, nickvl@umich.edu), Jonmarcos Diaz, and Kevin Maki (Univ. of Michigan, Ann Arbor, MI)

High-Reynolds number turbulent boundary layers are an important source for inducing structural vibration. Small geometric features of a structure can generate significant turbulence that result in structural vibration. In this work, we develop a new method to couple a high-fidelity fluid solver with a dynamic hybrid finite element formulation for the structure. The plate where the flow excitation is applied is considered as part of a larger structure. A hybrid approach based on the Component Mode Synthesis (CMS) is used for developing the new hybrid formulation. The dynamic behavior of the plate which is excited by the flow is modeled using finite elements. However, the rest of the surrounding structure is modeled using finite elements for the static modes and an analytical solution for the

dynamic modes of the CMS decomposition. The two main elements of the new work, the hybrid structural formulation and the process of applying the fluid load on the structural dynamic model are discussed. Validation of the new methodology is done by using test data from the literature for the vibration of plates excited by air flow, and through comparisons between the new methodology and traditional finite element based solutions.

9:50

2aSA4. Measurement of turbulent boundary layer unsteady wall pressures beneath elastomer layers of various thicknesses on a plate. Cory J. Smith (Penn State Univ., 1109 Houserville Rd., State College, PA 16801, coryjonsmith@gmail.com), Dean Capone, and Timothy A. Brungart (Penn State Univ., State College, PA)

The attenuation of turbulence induced wall pressure fluctuations through elastomer layers was studied experimentally and analytically. Wall pressure statistics were measured downstream from a backward facing step, with no elastomer present and beneath 2, 3, and 4 mm thick elastomers in a water tunnel facility. The step height, h, was 0.635 cm, and the wall pressures were measured at non-dimensional distances of $x/h = 10$, 24, 36 and 54 downstream from the step. The attenuation of the wall pressure spectra beneath the elastomer layers that was measured experimentally was then compared to analytical model predictions. An analytical elastomer transfer function, which models the transfer of turbulent boundary layer wall pressures on the surface of an elastomer to the normal stresses through the elastomer, was applied to the turbulent boundary layer wall pressure spectra measured in the absence of an elastomer layer and compared to spectra measured beneath the 2, 3, and 4 mm thick elastomers. The attenuation of the turbulent boundary layer wall pressure spectra through the elastomer layer using the analytical elastomer transfer function was in excellent agreement with the attenuation measured experimentally through all thicknesses of elastomer and at all free stream velocities at which the experiments were performed.

10:10–10:25 Break

10:25

2aSA5. Wavenumber scattering and the interactions of turbulent boundary layer flow with the structures exhibiting the acoustic black hole effect. Micah R. Shepherd (Appl. Res. Lab., Penn State Univ., PO Box 30, mailstop 3220B, State College, PA 16801, mrs30@psu.edu), Philip A. Feurtado, and Stephen C. Conlon (Appl. Res. Lab, Penn State Univ., State College, PA)

Beam and plate-like structures with non-constant surface impedance are known to exhibit wavenumber scattering such that the wavenumber energy of a vibration mode of the structure is spread over a wide wavenumber band. The scattered energy can couple into partially correlated forcing functions, such as turbulent boundary layer (TBL) flow, which exhibit a broadband wavenumber spectrum centered at the convective wavenumber. The overall effect is an increased amount of radiated noise for certain flow speeds. Recently, structures with imbedded acoustic black holes (ABH) have shown good noise and vibration level reductions for point excitations. However, it will be shown that when excited by complex, partially correlated forcing functions such as TBL flow, the wavenumber scattering caused by the power-law taper of the ABH can lead to increased radiated noise levels. The flow speed and wavenumber dependence of the TBL/ABH interactions will be presented using dimensional analysis.

Contributed Papers

10:45

2aSA6. Viscous flow sensing using micro-beam arrays. Mahdi Farahikia (Mech. Eng. Dept., SUNY Binghamton, Rm. 1330, 85 Murray Hill Rd., Binghamton, NY 13902, mfarahi1@binghamton.edu) and Ronald Miles (Mech. Eng., SUNY Binghamton, Vestal, NY)

Thin, compliant fibers have been shown to provide a remarkably accurate means of detecting sound [Zhou *et al*., "Sensing fluctuating airflow with spider silk," *Proc. Natil. Acad. Sci*., 201710559 (2017)]. The present study extends these results by examining the forces due to air-borne sound on a periodic array of infinitely long and thin micro-beams having rectangular cross section. Results obtained using both the Finite Element Method and analytical solutions to Stokes' equations are shown to be in excellent agreement. It is found that viscous forces are dominant when the width of the beams is less than 10 μm. Unlike typical MEMS resonators, which perform best in the absence of air, the dominance of viscous damping forces is desirable in directional acoustic flow sensors using these micro-beams. Furthermore, the frequency independent characteristic of these forces promises a flat response across a wide range of frequencies. It is predicted that smaller beams with wider gaps provide better sensitivity. For beams that are sufficiently thin, their thickness is found to have minimal influence on their performance.

11:00

2aSA7. An initial analysis of two western rivers' turbulent-flow-induced noise. James R. Brady (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84604, jebrady00@gmail.com), Kent L. Gee, and Mark K. Transtrum (Phys., Brigham Young Univ., Provo, UT)

This talk describes noise characteristics near turbulent sections of the Provo and Colorado Rivers in Utah. For reference, these rivers are characterized as medium-sized bedrock rivers, with Strahler orders of 5 and 7, respectively. Larson Davis 831C sound level meters were used to obtain two-second, one-third octave time histories within several meters of the bank at different locations and different times of day. Along the Provo River, the low-frequency content (below 250 Hz) varies with time of day and is likely contaminated by anthropogenic noise sources. The high-frequency noise (above 250 Hz) spectral shape is remarkably similar for every measurement; this spectrum is possibly characteristic of turbulent flow in smaller rivers. However, a measurement near a Colorado River Class III rapid reveals significantly more low-frequency spectral content, but similar high-frequency behavior. Initial efforts to correlate noise with available physical data on both river systems such as flow rates, river depths, river gradient, and terrain composition are described.

11:15

2aSA8. Impact of airfoil design uncertainty on the prediction of gust response. Gage S. Walters (Appl. Res. Lab., Penn State Univ., P.O. Box 30, M.S. 3220B, State College, PA 16804, gvw5074@psu.edu), Andrew S. Wixom, Sheri Martinelli, Zachary P. Berger, Michael H. Krane, Amanda Hanford, and Peter Lysak (Appl. Res. Lab., Penn State Univ., State College, PA)

Unsteady lifting forces acting on an airfoil in turbulent flow can be an important source of vibration and noise in many applications. Models have been previously developed for predicting the unsteady lift forces that account for the design and shape of the airfoil, in particular showing that leading-edge thickness, camber, and angle of attack can create large differences in the high-frequency decay of the unsteady lift force. Small changes in these design parameters can have a profound effect on the response of the airfoil that can lead to increased noise and vibration levels. This study examines the variability of the unsteady lift due to uncertainty in such parameters using an uncertainty quantification technique known as generalized polynomial chaos. Probability distributions are assumed for each of the input parameters in order to calculate the overall stochastic behavior of the response as well as its sensitivity to each input. The resulting predictions are compared against experimental measurements.

11:30

2aSA9. Incorporating acoustic black holes in hydrofoils. Kaushik Sampath (Code 7165, NRC Postdoctoral Associate, U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-0001, kaushik.sampath.ctr.in@nrl.navy.mil), Caleb F. Sieck, Matthew D. Guild, and Charles Rohde (Code 7165, U. S. Naval Res. Lab., Washington, DC)

Fluid-loaded structures such as turbine blades, ship hulls, and aircraft wings are required to withstand turbulent flow-induced structural vibrations and yet be lightweight. In this study, the feasibility of applying the concept of acoustic black holes (ABHs) into hydrofoils as a means to mitigate flow-induced structural vibrations is evaluated. ABHs are geometrical tapers that “trap” flexural vibrations near an ideal edge. Practical implementations of this approach involve the use of damping material near the truncated edge. Optimization of the ABH thickness profile, number of ABH elements, and the damping layer for mitigating surface vibrations is performed using finite element simulations. Select designs are then fabricated leveraging multi-material polyjet 3-D printing technology. Surface vibrations of the test samples subject to leading-edge excitations are characterized using laser Doppler vibrometry. Test results are compared with simulation data and a design framework/tool for ABH implementation in hydrofoils is formulated. [Work sponsored by the Office of Naval Research.]

11:45

2aSA10. Numerical simulation of flow-induced sound from a wall-mounted finite length cylinder. Brett Lenz (Iowa State Univ., 1423 S Grand Ave., Rm. 303, Ames, IA 50010, blenz@iastate.edu), Jose F. Magalhaes, and Sanghoon Suh (Deere & Co., Moline, IL)

A study is conducted on the numerical simulation of flow-induced sound of a wall-mounted finite length cylinder. A circular cylinder with a length-to-diameter ratio of 22.6 was considered at a Reynolds number of 11 700. The flow field is computed using a hybrid LES/RANS model and the far-field noise is calculated using the Ffowcs Williams and Hawkings acoustic analogy. The flow and acoustic results are compared with experimental results published by Moreau and Doolan in 2013.

Session 2aSC

Speech Communication: Second-Language Acquisition and Bilingualism (Poster Session)

Matthew Faytak, Chair
University of California, San Diego, Cognitive Sciences, La Jolla, California 92093

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 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

2aSC1. Effects of language proficiency in categorical perception of Mandarin Chinese tones. Bing Cheng (Confucius Inst., Univ. of Nebraska-Lincoln, 900 N. 16th St., Nebraska Hall W205, Lincoln, NE 68588, bcheng3@unl.edu), Wen Zhang (English Dept., School of Foreign Studies, Xi'an Jiaotong Univ., Xian, Shaanxi, China), and Yang Zhang (Dept. of Speech-Language-Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

This study compared the perception of Mandarin Chinese lexical tones by 20 adult native Mandarin Chinese speakers and 17 Chinese as a foreign language (CFL) learners with different levels of Chinese proficiency. We were particularly interested in confirming the effects of language experience and further assessing how Chinese proficiency might influence the perception of Mandarin lexical tones. Three types of continuum were used in identification and discrimination tasks, including tone 1 (T1)/tone 2 (T2), tone 1/tone 4 (T4) and tone 2/tone 4. Unlike the native speakers of Chinese who demonstrated clear categorical perception for all the continua, the CFL learners showed reduced accuracy in identification and discrimination results. Furthermore, proficiency data correlated with categorical perception (CP) of Mandarin lexical tones, suggesting that categorical perception of key phonological contrasts in the second language can be considered as a reliable indicator of language proficiency.

2aSC2. Bilingual sibilant acoustics in conversational Cantonese-English speech. Khia A. Johnson (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, khia.johnson@ubc.ca) and Molly E. Babel (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

Similar sibilants in different languages can differ in their trajectories, even when other cues fail to differentiate them [Reidy, *J. Acoust. Soc. Am.* **140**, 2518 (2016)]. Whether bilinguals—whose languages influence each other in a multitude of ways—would maintain such a difference in sibilant production across their two languages remains an open question. Using the transcribed and force-aligned (female) half of a new corpus recorded in Vancouver, Canada during 2018–2019, this study compares word-initial, prevocalic Cantonese /s/ productions to English /s/ and /ʃ/ productions in the same environment, by the same set of talkers. Cantonese /s/ is of interest given its variable description in the literature, across measurements, talkers, and vowel environments [Yu, *J. Acoust. Soc. Am.* **139**, 1672 (2016)]. This study addresses how bilinguals produce Cantonese /s/, and whether or not it is acoustically comparable to either of their English voiceless sibilants. Sibilant productions are measured in a variety of ways, for comparison with previous studies. This includes peak ERB_N number trajectories, which capture spectro-temporal variation. All comparisons are within-talker. Cross-talker differences in language background are also considered.

2aSC3. Lexically guided perceptual learning in Cantonese-English bilinguals: A web replication study. Leighanne Chan (Commun. Sci. and Disord., Univ. of Western Ontario, London, ON, Canada), Khia A. Johnson (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, BC V6T 1Z4, Canada, khia.johnson@ubc.ca), and Molly E. Babel (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

Speech is incredibly variable, yet listeners have little difficulty adapting to new talkers. One proposed mechanism for how listeners rapidly map novel variants to established categories is *perceptual learning*. This study is a web replication of our previous work, which implements a version of perceptual learning that leverages lexical knowledge to retune phonetic categories [Norris *et al.*, *Cogn. Psychol.* **47**, 2014 (2003)]—here, Cantonese [f]. Embedded in a lexical decision task, Cantonese-English bilingual participants heard words where [f] was expected (e.g., 豆腐 *dau6fu6* "tofu"), but replaced with an ambiguous [f]-[s] sound. Participants then categorized tokens from ambiguous nonword-nonword continua. Lab participants in the experimental condition successfully retuned Cantonese [f], compared to controls. Replicating this finding online demonstrates the viability of the paradigm outside the lab for this population, and provides precedent for future work. By recruiting from the same population as our lab study, we can more directly compare lab and web results than previous studies with Amazon's Mechanical Turk. This provides a clearer picture of how the participants' environments drive variability in online speech perception research.

2aSC4. The perceptual assimilation and discrimination of 20 Hindi consonants by native speakers of English. Shannon L. Barrios (Linguist, Univ. of Utah, 255 S.Central Campus Dr., Rm. 2313, Salt Lake City, UT 84108, s.barrios@utah.edu) and Rachel Hayes-Harb (Linguist, Univ. of Utah, Salt Lake City, UT)

Hindi consonant contrasts are known to pose difficulties for native English speakers (e.g., Polka, 1991; Werker and Tees, 2002; and Cibelli, 2015). While a limited set of Hindi consonants have received a great deal of attention (i.e., the coronal stops), we do not yet know how learners perceive other segments from Hindi's relatively large consonant inventory. To address this gap, we conducted a perceptual assimilation study modeled after Faris *et al.* (2018) to investigate the perceptual assimilation of twenty Hindi consonants (tʃ, tʃʰ, dʒ, dʒʱ, t̪, t̪ʰ, d̪, d̪ʱ, ʈ, ʈʰ, ɖ, ɖʱ, ʃ, ʂ, s, z, l, r, ɽ, ɽʱ) by native English speakers with no prior Hindi language learning experience. We later examined the discrimination of 46 pairs from the same set of phones in a new group of participants from the same population using an AX discrimination

task. Participants exhibited patterns of perceptual assimilation and discrimination of Hindi phonemes that suggest that native English speakers will experience difficulty across the Hindi consonant inventory. We consider the findings in the context of the Perceptual Assimilation Model (Best, 1994 and Best and Tyler, 2007), providing a fuller account of the difficulty posed by the Hindi consonant inventory for second-language learners.

2aSC5. A quantitative model of talker normalization by native and non-native speakers. Si Chen (Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong, qinxi3@gmail.com), Caicai Zhang (Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), Puiyin Lau (Dept. of Statistics and Autuarial, Univ. of Hong Kong, Hong Kong, Hong Kong), Yike Yang (Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., Kowloon, Hong Kong), Bei LI (Chinese and Bilingual Studies, The Hong Kong Polytechnic Univ., Hong Kong, Hong Kong), and Wing Kam Fung (Dept. of Statistics and Autuarial, Univ. of Hong Kong, Hong Kong, Hong Kong)

Talker variability affects native and non-native speakers in speech perception of segmentals (e.g., Bent *et al*., 2010) and suprasegmentals (e.g., Wong and Diehl, 2003). Zhang and Chen (2016) reported that gender-specific F0 range may contribute significantly to Cantonese tone perception. However, a full understanding of how the population F0 distribution affects tone identification is missing. This study aims to bridge this gap by modelling tone distributions and testing how deviated distribution parameters affect tone identification by native and non-native speakers. Statistical modeling of a Cantonese speech corpus of 68 speakers showed that F0 values of three Cantonese tones follow skew-normal distributions with three parameters: location, shape, and scale. We proceeded to conduct two experiments with 28 Cantonese and 28 Mandarin listeners using both naturally produced tones by 34 Cantonese speakers and manipulated tones with F0 values generated from simulated distributions. A multinomial mixed effects model revealed significant main effects of location and shape parameters. Locally weighted scatterplot smoothing curves also differed dramatically between native and non-native listeners, indicating an effect from long-term F0 distribution representations on tonal identification. The results thus offer useful insights about how parametric representations of phonetic distributions is used in tone identification.

2aSC6. "No" versus "Aniyo": Back vowel diphthongization in heritage Korean. Andrew Cheng (Dept. of Linguist, Univ. of California, Berkeley, Berkeley, CA 94720, andrewcheng@berkeley.edu)

The effects of a second language on a speaker's first language phonology are well documented (e.g., Chang, 2013), and much has been done work specifically for heritage language speakers (Polinsky and Kagan, 2007; Montrul, 2010, 2015), who may be bilingual to some degree but are dominant in the majority language. Bilingualism research has identified "separate but nonautonomous phonological systems" (Paradis, 2001) in bilingual children, but few studies address heritage speakers' phonological systems in relation to ongoing sound change (Konopka and Pierrehumbert, 2010). The current study examines F1 and F2 of the back vowels /o/ and /u/ in samples of read Korean speech from a population of Korean American Californians who identified as natively bilingual and/or heritage speakers of Korean. Forty speakers read short conversational passages in Korean during a bilingual interview. Preliminary results indicate that the heritage speakers' /u/ vowel shows influence from California English, with greater F2 on average than their central /a/ vowel. The /o/ vowel also has a much lower F1 than expected given Korean's canonical five-vowel system. Ongoing analysis examines the trajectory of both vowels to determine the influence of California English diphthongized back vowels on Korean monophthongs, and accent perception of the speakers' Korean.

2aSC7. The influence of lexical characteristics and talker accent on the recognition of English words by speakers of Korean. Minkyoung Choi (Speech Lang. Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, choix690@umn.edu), Jeffrey J. Holliday (Dept. of Korean Lang. and Lit., Korea Univ., Bloomington, IN), and Benjamin Munson (Speech Lang. Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

A number of recent studies have examined whether English-language learners (ELLs) and native speakers of English (NS) show the same effects of lexical (word frequency [WF] and phonological neighborhood density [PND]) and indexical (speaker accent) factors on word recognition accuracy and speed. The motivation behind this work is to understand how different types of linguistic knowledge emerge over the course of learning an L2. Two recent studies, Imai *et al*. (2005) and Yoneyama and Munson (2017) used nearly identical methods and stimuli with two groups of ELLs: those whose first language (L1) was Spanish (Imai *et al*.) and those whose L1 was Japanese (Yoneyama and Munson), and found very different effects of PND on word-recognition accuracy. In this project, we use the methods reported by Imai *et al*. and stimuli on a third group of ELLs, those who L1 was Korean. In a preliminary report on this project (Choi *et al*., 2018), we found evidence that the Korean L1 ELLs performed more similarly to the Japanese L1 ELLs in Yoneyama and Munson. However, these conclusions were based on data from a small group of relatively high-proficiency ELLs. In the current study, we report the results of a larger group of ELLs, including lower-proficiency individuals.

2aSC8. The impact of language transfer on native speaker recognition of native and non-native speech. John Culnan (Dept. of Linguist, Univ. of Arizona, Tucson, AZ 85721, jmculnan@email.arizona.edu) and Seongjin Park (Univ. of Arizona, Tucson, AZ)

There are many identifiable acoustic differences between native and non-native English, including patterns of prosody (Ramirez Verdugo, 2005), vowel quality (Flege *et al*., 1997) and extent of reduction in speech (Baker *et al*., 2011), many of which arise due to transfer from the native language (L1 transfer). This study examines the effect that differences generated through language transfer have upon the recognition of the Mandarin-accented English compared to American English by native speakers of American English. Listeners were presented with a set of 40 different TIMIT sentences (Garofolo *et al*., 1993) spoken by 5 native speakers of American English and 5 native Mandarin speakers, and were asked to transcribe what they heard. These transcriptions will be evaluated for accuracy, with errors identified and classified according to the type(s) of acoustic deviations from native English speech attested in any misheard words, as well as whether these errors are typical of Mandarin-speaking learners of English. Results will be used to evaluate the relative importance of these differences between native and non-native English speech in recognition and to identify potentially problematic speech patterns for use with multi-dialectal systems of automatic speech recognition.

2aSC9. Effects of speech-shaped noise on consonant identification by native and non-native Korean and English listeners. Minkyoung Choi (Speech Lang. Hearing Sci., Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr. SE, Minneapolis, MN 55455, choix690@umn.edu), Benjamin Munson, and Evelyn E. Davies-Venn (Speech Lang. Hearing Sci., Univ. of Minnesota, Minneapolis, MN)

The global proliferation of machine-human interface via voice and audio suggests a growing need for including language-specificity in the design and implementation of acoustic signal processing devices for all individuals, especially those in educational settings. Second language learners are often

more susceptible to acoustic distortions in their second language (i.e., L2). Speech understanding challenges that L2 learners face in noisy environments are often compounded by other normal cross-language influences that are expected to occur during L2 speech perception. This study evaluated consonant confusion patterns using consonant-vowel (CV) English tokens spoken by native Korean and English speakers. The speech stimuli were embedded in speech-shaped noise with varying signal-to-noise ratios. Speech scores were measured using an alternative forced-choice paradigm. Overall scores were evaluated for each listener group to first assess whether L2 learners were most robust to acoustic-variances in English when spoken by Korean compared to English native-speakers. Second, their speech scores and acoustic recordings were evaluated for constructive as well as destructive effects of these acoustic variance on their masking and consonant confusion patterns. Results will be presented on the similarity and differences in acoustics and perception between English and Korean native-speakers. The clinical implications of our findings will be discussed.

2aSC10. Assessing the differences between monolingual and bilingual speakers in a tongue-twister task: Is there evidence for a bilingual advantage? Beckie D. Dugaillard (Commun. & Performing Arts, City Univ. of New York - Kingsborough Community College, 503 Vermont St., Brooklyn, NY 11207, beckie.dugaillard11@students.kbcc.cuny.edu), Mariana Vasilita (Communications & Performing Arts, City Univ. of New York - Kingsborough Community College, Brooklyn, NY), and Laura Spinu (Communications & Performing Arts, City Univ. of New York - Kingsborough Community College, Brooklyn, NY)

While not uncontroversial, the claim that bilingual experience underlies certain cognitive advantages has been at the forefront of recent work. Thus, enhanced skills were associated with bilingualism in terms of multitasking, auditory encoding of sound, and resistance to dementia. Among the mechanisms potentially supporting the cognitive differences between mono- and bilinguals, those pertaining to executive function have been investigated extensively. Our study adds to a more recently initiated direction seeking to determine if there is also a connection between bilingualism and sensorimotor mechanisms. Specifically, we explore experimentally articulatory skill, expressed as accuracy, speed, and acoustic prototypicality in the production of tongue-twisters. Three groups of undergraduates (n = 40)—monolingual, early bilinguals, and late bilinguals—read artificially constructed tongue-twisters such as "kifkivkivkif" three times in succession, matching a 150 bps rhythm on a metronome. After a practice session, each participant read 64 items, many of which induced mispronunciations. The productions were manually evaluated for correctness, revealing that the monolinguals and late bilinguals outperformed the early bilinguals. In the next phase, acoustic analyses will be performed to determine more fine-grained phonetic properties, e.g., consonant duration, formant transitions, and extent of devoicing, possibly revealing different hyper-/hypoarticulation patterns in the three groups.

2aSC11. The identification of trained nonnative speech sounds improve after a daytime nap. F. Sayako Earle (Commun. Sci. and Disord., Univ. of Delaware, 100 Discovery Boulevard, Newark, DE 19713, fsearle@udel.edu)

Perceptual ability of nonnative speech sounds improves in the 24 hours that follow learning, in the absence of further training (Earle and Myers, 2015a, 2015b). These improvements are particularly salient in the morning, that is, immediately following the overnight between-session interval. These observations are moreover associated with rudimentary measures of sleep duration (Earle *et al.*, 2017). While we have attributed these perceptual gains to memory consolidation during sleep (Marshall and Born, 2007), there yet remains the possibility that consolidation of acoustic information occurs during a relative period of absence of interfering auditory input that is related to time spent in sleep. In this study, we trained a group of college students in the identification of the Hindi dental-retroflex contrast at noon, and then recorded their electroencephalographic (EEG) activity during an in-lab nap immediately following training. We measured perceptual performance on the trained nonnative contrast immediately before and after a daytime nap. Data from a preliminary sample of 8 participants demonstrate a trending association between changes in perceptual behavior and the number of sleep spindles observed during stage N2 post-training sleep. This association may indicate the role of sleep, and hippocampal memory consolidation, in the formation of speech sound representations.

2aSC12. Spanish and English bilinguals: Between L1 and L2? Aline Ferreira (Spanish and Portuguese, UCSB, 505 Storke Rd., Apt. 7311, Goleta, CA 93117, aferreira@spanport.ucsb.edu) and Viola G. Miglio (Spanish and Portuguese, UCSB, Santa Barbara, CA)

The present study explores the vowel and intonation systems of heritage Spanish speakers in California based on a corpus of semi-directed interviews, to establish whether one language influences the other, or whether their English and for Spanish are closer to the monolingual speakers' system of either language. There is some limited evidence that the vowel systems of Spanish in Spanish heritage speakers of Mexican descent in California is indeed very similar to the system of monolingual Mexican Spanish speakers (in pitch, duration, F1 and F2 values)—their English vowel system, however, and especially the front vowels, is not completely comparable to the published values for monolingual American English speakers' vowels (Miglio, 2011). Intonation in bilingual Spanish has been shown to be influenced by English pitch movement, as well as signalling information structure (Gries and Miglio, 2015). This study widens the scope of the previous ones using data from natural speech as opposed to carrier sentences read out loud, and uses unpublished data from the same regional varieties of both English and Spanish, as spoken in California (monolingual English, bilingual English-Spanish), and in Mexico (monolingual Spanish) by individuals in the same age group and of comparable socio-cultural background.

2aSC13. Geminate attrition in the speech of Arabic–English bilinguals living in the United States. Rawan Hanini (Speech Commun., Kingsborough Community College, 1149 72nd St., Brooklyn, NY 11228, rawanhanini@hotmail.com), Anwar Khudidi, Yasaman Rafat (Dept. of Modern Lang. and Literatures, The Univ. of Western ON, London, ON, Canada), and Laura Spinu (Speech Commun., Kingsborough Community College, Brooklyn, NY)

This study explores the phenomenon of language attrition. Specifically, we investigate the phonetic properties of consonant gemination across three groups of speakers of Palestinian Arabic: *monolinguals* (i.e., native speakers born in Palestine who have lived there their entire life, n = 5), *late bilingual*s (i.e., speakers born in Palestine who emigrated to the US during their teens, n = 6), and *heritage speakers* (i.e., speakers of Palestinian descent, born in the US and who speak both English and Arabic in their daily lives, n = 7). All speakers were in their mid-1920s. The participants were tested using a delayed word repetition task. The stimuli comprised 158 bi-syllabic Arabic minimal and near-minimal pairs (e.g., /ħam:aːm/ "bathroom" versus /ħama:m/ "pigeon") including long and short stops, fricatives, and sonorants. We controlled for stress and syllabic position. Distractors were also included. The acoustic analysis is underway, and consists of manually aligning the target consonants, extracting the mean consonant duration and comparing it across groups. Additional measures include voicing, aspiration, and formant transitions. The findings will enable us to address the question whether universal phonetic factors (from the perspective of Markedness Theory) have an effect on degree of attrition by specifically comparing consonants from different voicing categories and manners of articulation.

2aSC14. Effects of L1 on the voicing of intervocalic voiceless stops and its relation with L2 proficiency. Ji-Hyun Jeon (Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul KS013, South Korea, wjswlgus6623@hanmail.net), Jiyun Yoo (Yonsei Univ., Seoul, South Korea), and Seok-Chae Lee (Yonsei Univ., Seoul, Sedeamum-Gu, South Korea)

This study investigates the correlation between the proficiency of L2 learners of English and the degree of L1 Korean intervocalic voicing rule transfer on the English intervocalic voiceless stop production by Korean English learners. Korean has a phonological rule that voiceless stops become voiced between vowels (Korean intervocalic voicing rule) but this rule does not apply in English. The aforementioned difference between the phonological rule of the two languages makes it possible to predict that

Korean learners with lower L2 proficiency might produce English intervocalic voiceless stops with higher degree of voicing than the learners with higher L2 proficiency. The speech data of Korean L2 learners of English is extracted from GenieSpeeCor, rated and categorized English speech corpus of Korean L2 learners of English production, and the value of intensity drop and closure duration were used as acoustic cues for voicing difference. Alveolar stop is excluded from the data analysis because the flapped /t/ sound is difficult to distinguish from the voiced sound. As a result of the current study, when it comes to the closure duration as an acoustic cue, there was no meaningful statistical relation between learners' L2 proficiency and the degree of intervocalic stops voicing. However, in terms of the intensity drop learners with the highest L2 proficiency showed a lower degree of voicing than the learners with middle and lowest proficiency.

2aSC15. Effects of language experience and sentential context in categorical perception of Mandarin tones. Siying Fan (English, West School of Foreign Studies, Xi'an Jiaotong Univ., 28 Xianning St. Xi'an, Shaanxi 710049, China, fancy4869@stu.xjtu.edu.cn) and Bing Cheng (English, Xi'an Jiaotong Univ., Xi'an, Shaanxi, China)

Previous research shows that sentential context plays an important role in lexical tone recognition when the fundamental frequency is degraded by noise or flattened. The present study aimed to test how language experience and sentence context would jointly influence the categorical perception of Mandarin tones. Thirty native Chinese speakers and thirty CFL (Chinese-as-a-foreign-language) learners participated in the study. The experimental protocol used identification and discrimination tasks for categorical perception of the synthetic tone2-tone3 continuum. Experimental conditions included the target tones in isolation and the same tones embedded in initial, medial and final position of sentences. The location, slope, and width of the phonetic boundary in identification function and the discrimination accuracy were assessed. The categorical perception test results indicated strong influences of language experience and sentential context, which has important pedagogical implications for second language learners.

2aSC16. Knowledge of word stress patterns in English in Korean Learners: The case of English trisyllabic nouns. Sung Yeon Kim (Dept. of English Lang. and Lit., Ewha Womans Univ., 52 Ewhayeodae-gil, Daehyeon-Dong, Seodaemun-Gu, Seoul 03760, South Korea, joansy@ewhain.net) and Eunjin Oh (Dept. of English Lang. and Lit., Ewha Womans Univ., Seoul, South Korea)

This study aims to explore Korean learner knowledge of word stress patterns in English trisyllabic nouns. Building upon the previous findings from Guion (2005), further studies should look at data regarding more syllabically complex words. While the stress placement rules in English require at least three syllables to be applied, the study (Guion, 2005) only observed the individuals' stress placement patterns in English disyllabic words. The prevalence of longer polysyllabic words in the English language necessitates observation of Korean learner speech patterns in English trisyllabic words. A total of 16 Korean learners of English participated in the experiments that explore the distributional patterns of stress placement based on word type (i.e., nonwords and real words) and syllable stress position on English trisyllabic nouns. Compared to that of native English speakers, Korean learners of English showed significantly higher error rate on stress placement rules in English nouns—particularly higher error rates on production and perception of nonwords, compared to that of English real words. The predictive factors (word type, syllable structure, and syllable weight) had a significant effect on the stress placement on both groups. Based on the results, our goal is to discuss how the learners use their previous language learning experience when they decide on which syllable to apply stress.

2aSC17. Non-native English speakers' adaptation to native English speaker's speech. Dae-yong Lee (Dept. of Linguist, Univ. of Oregon, 161 Straub Hall, 1290 University of Oregon, Eugene, OR 97403, daeyongl@uoregon.edu) and Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, Eugene, OR)

Previous studies have shown that listeners are able to quickly adapt to unfamiliar speech (e.g., regional-accented, foreign-accented, or time-compressed speech). While listeners initially have difficulty understanding unfamiliar speech, they are able to improve their ability to understand speech in a short period of time. Prior studies that examined adaptation to unfamiliar speech mostly focused on adaptation by native listeners, and it is less clear whether non-native listeners are able to adapt to native speech. This study aims to examine whether non-native speakers of English adapt talker-specifically to native English speech. Further, this study also examines, if non-native English speakers do adapt to native English speech, whether the intelligibility of utterances affects the adaptation. In the present study, non-native English speakers listen to English sentences read by native English speakers and determine whether the last word they heard matches the word on the screen. Response time is measured to examine whether non-native English speakers adapted to native English speaker's speech and whether the intelligibility of speech interacts with the adaptation. The results of the study will provide a better understanding of non-native English speakers' adaptation to native English speech and help improve the efficiency of conversations between native and non-native speakers.

2aSC18. The study of the relationship between Koreans' English proficiency and vowel insertion after word-final stops in English. Hyunjun Lee (English Lang. & Lit., Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, hjne95@yonsei.ac.kr), Wooji Park (English Lang. & Lit., Yonsei Univ., Seoul, South Korea), and Seok-Chae Lee (English Lang. & Lit., Yonsei Univ., Seoul, Sedeamum-Gu, South Korea)

Vowel insertion is one of the utterance errors that Korean learners of English commit frequently. Based on this fact, the purpose of this research is to determine whether the Koreans' English proficiency affects the phenomenon of vowel insertion. To achieve it, the study investigated the relationship between the Korean speakers' English proficiency and vowel insertion in their English read speech. For analysis, Korean-Spoken English Corpus (K-SEC, made in 2004) and rated K-SEC (made in 2017) were used where 32 Korean elementary school students from Seoul and Gyeonggi province had pronounced the 6 target items that are minimal pairs (e.g. cap-cab). The speakers in these speech corpus data were also classified in one of the three different groups according to their English proficiency—Novice, Intermediate, and Advanced. It turned out that the speakers who are in the advanced level showed a tendency of the least chance of inserting a vowel after word-final voiced stops (7%) whereas 35% of the speakers who are in the novice proficiency level did so. For voiceless stops, the result was similar; the advanced group had the lowest tendency of inserting a vowel after word-final voiceless stops (4%) while novice group made 18% of vowel insertion. From these finding, it is concluded that the more proficient the Korean speakers in English, the less they are likely to insert a vowel after word-final stops.

2aSC19. The effect of clear speech and masking noise on listening effort in native and non-native listeners. Kirsten Meemann (Linguist, Univ. of Texas at Austin, 305 E. 23rd St., Mail Code B5100, Austin, TX 78712, kirsten.meemann@utexas.edu) and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Austin, TX)

Listening effort increases when perceiving speech masked by noise and when listeners are non-native speakers of the target language (Zekveld and Kramer, 2014; Borghini and Hazan, 2018). Listener-oriented clear speech

(CS) improves word recognition in noise and recognition memory (Keerstock and Smiljanić, 2018). It is unclear, however, whether the CS benefit can be explained by reduced listening effort during processing of easier-to-understand speech. The goal of this study was to examine the effect of CS on listening effort associated with degraded listening conditions differing in the source of difficulty and underlying cognitive mechanism: energetic masking, informational masking, and non-native listening. Native and non-native English listeners performed a word recognition in noise task and a visual response task, i.e., pressing keys to indicate the orientation of an arrow on the screen, first separately and then simultaneously. In the word recognition task, listeners repeated back sentences produced in CS and conversational speech and mixed with speech-shaped noise and 2-talker babble. The presentation will discuss analyses aimed at uncovering how hyper-articulated CS and masker type influence listening effort for native and non-native listeners as measured by objective and subjective dual-task cost.

2aSC20. The relation between the vowel duration and intensity in English NPs [an article—a monosyllabic noun] and the English proficiency of Korean learners. RAN MO (English Lit. and Linguist, Yonsei Univ., Seodaemoon-Gu, Yonsei-Ro 50, Seoul 04501, South Korea, 0728mr@gmail.com), Wooji Park, and Seok-Chae Lee (English Lit. and Linguist, Yonsei Univ., Seoul, South Korea)

Generally, the vowels in content words are phonetically more prominent than the ones in function words in English. This is because they have a stress of a suprasegmental feature, making them produced longer in duration and greater in intensity. Based on this phonetical phenomenon, this research investigates the relation between the vowel duration and intensity in English NPs [an article—a monosyllabic noun] and the English proficiency of Korean learners. For analysis, the researchers utilized Rated Korean-Spoken English Corpus (made in 2017) to collect and measure the duration and intensity of each vowel in the monosyllabic noun phrases of the 853 sentences articulated by the Korean speakers who were categorized into one of the four different English proficiency groups based on their speaking ability (Advanced, Intermediate-High, Intermediate-Low, and Novice). From the findings, it was revealed that the higher the proficiency level becomes, the greater the ratio of the vowel duration in the nouns of a content word to the vowel duration in the articles of a function word gets, meaning that the higher the English proficiency level reaches, the better the Korean speakers articulate the words with proper stresses as native speakers do. However, the ratio of the vowel intensity in the nouns to the vowel intensity in the articles has less relation to Korean speakers' English proficiency.

2aSC21. Effects of Individual difference and L1 phonology on the identification and discrimination of American English vowels by native Japanese and Korean listeners. Takeshi Nozawa (Lang. Education Ctr., Ritsumeikan Univ., 1-1-1 Nojihigashi, Kusatsu 525-8577, Japan, t-nozawa@ec.ritsumei.ac.jp)

Native speakers of Japanese and Korean identified and discriminated American English vowels /i, ɪ, ɛ, æ, ɑ, ʌ/ in four different consonantal contexts. These listeners also perceptually assimilated the American English vowels to their respective L1 vowel categories. In another session, these listeners produced /i, ɪ, ɛ, æ, ɑ, ʌ/ in two different conditions. They read aloud words on a list, and they repeated after native speakers' utterances. The results revealed that Korean listeners as well as Japanese listeners rely on durational difference to identify and discriminate /i/ and /ɪ/, but Korean listeners perceptually assimilated /i/ and /ɪ/ as roughly equally good exemplars of Korean high front vowel. Japanese listeners, on the other hand, perceived /ɪ/ as somewhat less ideal exemplar of Japanese high front vowel than /i/. However, those Japanese listeners who were sensitive to spectral differences did not outperform those who were less sensitive. Moreover, Japanese listeners sensitive to the spectral differences often misidentified /ɪ/ for /ɛ/. An MDS analysis revealed that in the identification task, listeners' L1 phonology exercise a stronger influence than do individual differences, but in discrimination task, individual differences are as strong a factor as listeners' L1 phonology.

2aSC22. The effect of word-final stops' voicing on the vowel duration and its relation with Korean speakers' english proficiency. Wooji Park (English Lit. & Lang., Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, woojipark@yonsei.ac.kr), Hyunjun Lee (English Lit. & Lang., Yonsei Univ., Seoul, South Korea), and Seok-Chae Lee (English Lit. & Lang., Yonsei Univ., Seoul, Sedeamum-Gu, South Korea)

It has been well known that the vowels preceding voiced stops are produced durationally longer than the vowels preceding voiceless stops in English. Based on this phonetic phenomenon, this research investigates to determine whether there is any relation between the Koreans' English proficiency and the difference in the ratio of the English vowel duration before the voiced stops to the English vowel duration before the voiceless stops. For analysis, the Korean-Spoken English Corpus (K-SEC, made in 2004) and the Rated Korean-Spoken English Corpus (Rated K-SEC, made in 2017) were utilized where the 32 Korean elementary school students from Seoul and Gyeonggi province articulated the target items such as bag/back and cab/cap. These speakers were categorized into one of the three different proficiency groups (Advanced, Intermediate, and Novice) according to their speaking ability. From the findings, native English speakers were found producing the vowels before the word-final voiced stops 1.421 times longer than the vowels before the word-final voiceless stops. Similarly, in the case of the Korean speakers, they produced the vowels before the word-final voiced stops longer than the vowels before the word-final voiceless stops. However, the study couldn't discover any possible relationship between their English proficiency and the ratio of the vowel duration before the word-final voiced stops to the vowel duration before the word-final voiceless stops in English (the ratio—Advanced: 1.275, Intermediate: 1.209 and Novice: 1.25).

2aSC23. Exploring the Lombard Effect in first language Japanese speakers of English. Saya Kawase (Faculty of Sci. and Eng., Waseda Univ., Tokyo, Tokyo-To, Japan), Michael L. Smith (Linguist, Univ. of Washington, 1417 NE 42nd St., Box 354875, Seattle, WA 98105, smithm59@uw.edu), and Richard Wright (Linguist, Univ. of Washington, Seattle, WA)

Speakers adjust pronunciations in noise to make communication more effective. This phenomenon, known as the *Lombard Effect* after Etienne Lombard who first described it in 1911, is characterized by increased intensity, elongation of vowels, emphasis on stress syllables, increased pitch range, and changes to the vowel space (see Brum and Zollinger, 2011). Because the Lombard Effect interacts with linguistic factors (e.g., Wassink *et al*., 2006 and Patel and Schell, 2008) we hypothesize that a speaker's native language will interact with the Lombard Effect and that second language speakers of English will carryover these effects. To test this prediction, we recorded native speakers of North American English and L1 Japanese speakers of L2 English participating in a modified map task to illicit natural communication between the speakers and the experimenter, in two conditions: noisy and quiet (presented over headphones). Measurements of vowel duration, vowel quality, and intensity were extracted from key words representing monosyllabic and trochaic words with six vowel qualities: /i ɪ a æ u ʊ/. Preliminary results indicate some language specific differences, but also unexpectedly that vowel space is more compressed in the Lombard condition than in quiet, which has connections to findings of Zhao and Jurafsky (2009) for Mandarin.

2aSC24. The acquisition of a novel accent by monolinguals and early bilinguals: Vowel epenthesis in English voiceless s-clusters. Mariana Vasilita (City Univ. of New York- Kingborough Community College, 2001 Oriental Blvd, Brooklyn, NY 11235, mariana.vasilita34@students.kbcc.cuny.edu), Beckie D. Dugaillard, and Laura Spinu (City Univ. of New York- Kingborough Community College, Brooklyn, NY)

Bilingualism has been linked with improved function regarding certain aspects of linguistic processing, e.g., manipulating language in terms of discrete units, novel word acquisition, and learning unfamiliar sound patterns in novel accents. Recent experimental work with non-native contrasts suggests that bilinguals have enhanced phonetic learning and speech perception abilities compared to monolinguals. We investigate phonetic learning skills in monolinguals (n = 20) and early, simultaneous bilinguals (n = 20). The subjects were trained and tested on an artificial accent of English. One the features distinguishing the novel accent from the standard variety was the presence of vocalic epenthesis in the voiceless s-clusters [sp, st, sk]. For example, words such as "spy" or "school" were pronounced as "suh-py" [səpaj] and "suh-cool" [səkul]. A total of 760 target items were evaluated manually. A score of 1 was assigned for each token produced with an epenthetic vowel between s and the following consonant, otherwise the score was 0. Early bilinguals outperformed the monolinguals on the acquisition of this pattern. Acoustic analyses are underway to assess the differences between the two groups' productions, such as quality, duration, and placement of the epenthetic vowel, as preliminary observations revealed a high number of long-distance phenomena such as [spəhaj] or [spəʔaj].

2aSC25. Examining vowel nasalization in English word productions of bilingual and monolingual Cajun heritage speakers. Lauren V. Vidrine (Louisiana State Univ., 5151 Highland Rd., Baton Rouge, LA 70808, lvidri5@lsu.edu) and Irina A. Shport (Louisiana State Univ., Baton Rouge, LA)

One perceived characteristic of Cajun English (CE) spoken in the state of Louisiana is vowel nasalization, which has been attributed to Cajun French (CF) influence and has been reported in oldest generation Cajuns and young generation Cajun men (DuBois and Horvath, 1998, 2000). Previous research involved only CF-CE bilinguals and sociolinguistic coding based on auditory impressions. In this study, acoustic measures were used to investigate vowel nasalization among both CF-CE bilinguals (the oldest generation) and CE monolinguals. Five bilinguals and five monolinguals completed a word-reading task. In 39 monosyllabic words, four vowels occurred in both oral and nasal environments (CVC, CV, CVN). Nasality measures included bandwidth of F1; the difference between F1's amplitude and the highest of the first or second harmonic (A1-P0); and the difference between F3's amplitude and P0 (A3-P0). These were identified as most effective to distinguish oral and nasal vowels (Styler, 2017). Results showed that in comparison to monolinguals, vowels produced by bilinguals were more nasalized in both oral and nasal environments. This suggests that the vowel nasalization may be fading in CE, as it is lessened among young monolinguals. The results are discussed in relation to theories regarding cross-linguistic influence in bilingual communities.

2aSC26. Calibrating rhythms in L1 Japanese and Japanese accented English. Ratree Wayland (Linguist, Univ. of Florida, 2801 SW 81st St., Gainesville, FL 32608, ratree@ufl.edu) and Takeshi Nozawa (Lang. Education Ctr., Ritsumeikan Univ., Kusatsu, Japan)

This study applies an automated procedure, Amplitude Envelope Modulation Spectrum (AEMS), to probe rhythmic differences between L1 Japanese and Japanese accented English. AEMS directly and automatically quantifies temporal regularities in the amplitude envelope of the speech waveform within specified frequency bands and has been shown to successfully differentiate types of dysarthria and utterances with and without code-switches. Ten native speakers of Japanese (1 male, 9 female) produced the rainbow passage in both Japanese and English. The passages will be segmented into sentences before AEMS application. The AEMS consist of the slow-rate (up to 10 Hz) amplitude modulations of the full signal and 7 octave bands ranging in center frequency from 125 to 8000 Hz. Six variables relating to peak frequency and amplitude and relative energy above, below, and in the region of 4 Hz will be calculated from each frequency bands. Discriminant function analyses (DFA) will then be performed to determine which sets of predictor variables best discriminate between the two passages.

2aSC27. Vowel reductions in the acquisition of English lexical stress by Kazakh-Russian bilinguals. Mahire Yakup (World Lang. and Lit., Nazarbayev Univ., Block 38-1104, 53 Kabanbay Batyr Ave., Astana, Aqmola 010000, Kazakhstan, yakefu.mayila@nu.edu.kz) and Zeinekhan Kuzekova (Deparment of Kazakh Lang. and Turkic Studies, Nazarbayev Univ., Nur-Sultan, Kazakhstan)

This article will focus on the vowel reductions in producing English lexical stress by Kazakh-Russian bilinguals. Even though previous research examined that Kazakh-Russian bilinguals produced English lexical stress using duration and intensity, but not fundamental frequency, using vowel reductions as cues is still a question. Since Russian and English as stress contrasted languages with vowel reductions; however, Kazakh is a fixed stress language without vowel reductions. The fourteen Kazakh-Russian bilinguals (IELTS $>$ 7.0) produced ten minimal pairs in English in the three contexts (in sentences, fixed conditions, and isolations). We used fixed conditions as samples for analysis. Since lack of American English native speakers as a baseline, we compared the stressed syllables versus unstressed syllables based on per item (PROject versus proJECT) by measuring F1/ F2 in both syllables. The results showed that participants used vowel reductions in a variety of ways. If there is schwa reduction, participants produced correctly in which F1 and F2 were changed according to the vowel positions. Non-schwa changes produced non-consistent results in which they manipulate F1 or F2 based on the vowels, and there are wrong reductions in which/ æ/produced as /ʌ/ in the example of "contract" as a final stressed word.

2aSC28. Temporal exaggeration facilitates second language phonetic training: The case of syllable-final nasal contrast. Bing Cheng (English Dept., School of Foreign Studies, School of Foreign Studies, Xi'an Jiaotong Univ., 28 Xianning St. West, Xi'an, Shaanxi 710049, China, bch@mail.xjtu.edu.cn), Xiaojuan Zhang (English Dept., School of Foreign Studies, Xi'an Jiaotong Univ., Xi'an, Shaanxi, China), and Yang Zhang (Dept. of Speech-language-hearing Sci. and Ctr. for Neurobehavioral Development, Univ. of Minnesota, Minneapolis, MN)

This study tested the generalizability and efficacy of a modified high-variability phonetic training (HVPT) method with temporal exaggeration. The target sounds were the syllable-final /n/-/ŋ/ contrast in English, which was found to be difficult for adult Mandarin Chinese speakers. Our training software program featured systematically controlled temporal exaggeration, multi-talker variability, audio-visual presentation, and adaptive listening. The participants were 24 Chinese college students, and they were randomly assigned to two groups. Pre- and post-tests were administered one week before and after training in the experimental group, including behavioral identification and discrimination of /n/-/ŋ/. The control group who did not receive training took the same tests. In addition, mismatch negativity (MMN) responses were also obtained from the training group to assess the training effects in within- and across- category discrimination without requiring focused attention. Compared with the control group, the training group showed enhanced categorical perception of the nasal contrast with corresponding changes in the MMN responses to within- and across- category differences. The data provide further evidence for incorporating temporal exaggeration in high variability phonetic training to promote brain plasticity at the perceptual and pre-attentive neural levels.

2aSC29. First-language effects on the production of Mandarin-accented English diphthongs: Insights from the perceptual assimilation model. Boram Kim (Linguist, CUNY Graduate Ctr., The Graduate Ctr., City Univ. of New York, 365 Fifth Ave., New York, NY 10016, bkim@gradcenter.cuny.edu), Wei-Rong Chen, D. H. Whalen (Haskins Labs., New Haven, CT), and Stefanie Reed (Linguist, CUNY Graduate Ctr., New York, NY)

Previous studies found that non-native vowel categorization can be explained by the Perceptual Assimilation Model (PAM), based on perception experiments. However, fewer production studies have been carried out to further support the findings. Here we examined English monophthongs and

diphthongs produced by Mandarin learners of English (MAE), compared with those produced by native American English (AE) Speakers, based on the articulatory and acoustic data in an electromagnetic articulometry (EMA) corpus [Berry and Johnson, 2014, ICASSP] with 20 MAE and 20 AE speakers. Monophthongs were examined at mid-point and diphthongs at nine equally-spaced time points (i.e., every 10% of a total vowel duration). Our preliminary results reveal that PAM-predicted perceptually assimilated vowel categories (i.e., vowel pair/contrast not exist in L1, such as the /i-ɪ/ contrast in AE) were indeed produced as a single category with large articulatory overlap. Further, if the English diphthong has a counterpart in Mandarin, it undergoes L1 transfer; if not, it is produced as a sequence of two monophthongs. For example, the English /aɪ/ can be categorized in Mandarin diphthong /aɪ/ but English /ɔɪ/ does not exist in Mandarin; and /aɪ/ produced by MAE underwent reduction as in native Mandarin diphthong but /ɔɪ/ did not.

TUESDAY MORNING, 3 DECEMBER 2019 REGENT, 8:00 A.M. TO 10:15 A.M.

Session 2aSP

Signal Processing in Acoustics: General Topics in Signal Processing II

Michael J. Bianco, Chair
Marine Physical Laboratory, University of California San Diego, Scripps Institution of Oceanography, 9500 Gilman Dr, La Jolla, California 92037

Contributed Papers

8:00

2aSP1. Unsupervised machine learning for acoustic tomography and event detection. Michael J. Bianco (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr, La Jolla, CA 92037, mbianco@ucsd.edu) and Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

We present acoustic tomography and event detection approaches based on unsupervised machine learning (ML). ML describes a set of techniques for automatically detecting and utilizing patterns in data. These techniques can roughly be categorized as either supervised or unsupervised learning. In supervised learning, an ML system (e.g., neural network (NN)) is trained to produce a desired output based on labeled training examples. In unsupervised learning, no labels are given and the task is to discover interesting or useful structure within the data. In acoustic and geophysical signal processing, often large training datasets are available, but have few labeled examples. We discuss how unsupervised ML can leverage training data without explicit labels to improve acoustic models. We first give an overview of the relevant ML theory. Next, we describe an approach to array tomography, called locally-sparse travel time tomography (LST). LST regularizes the inversion of travel time measurements for geophysical structure using dictionary learning (unsupervised). Dictionary learning is here used to obtain a dictionary of small-scale geophysical features that best represent the measurements. Finally, we discuss an approach to automated event detection in acoustic time series based on autoencoder NNs, which learn salient features from NN inputs with unsupervised learning.

8:15

2aSP2. Advanced automobile crash detection by acoustic methods. Yi Hang Sim (Electron. and Comput. Eng., Hong Kong Univ. of Sci. and Technol., Rm. 2448, Academic Bldg., Clear Water Bay, Hong Kong, yhsim@connect.ust.hk), Yijia Chen, Lijia Wu, Xianzheng Geng, Yuxuan Wan, and Kevin Chau (Electron. and Comput. Eng., Hong Kong Univ. of Sci. and Technol., Clear Water Bay, Hong Kong)

Today's automobiles are equipped with a variety of safety features. Onboard acceleration, rotation, steering angle, wheel speed, and throttle position sensors are widely incorporated in electronic stability control systems for traction and steering control. Cameras and even radars are increasingly employed for imminent crash detection and autonomous emergency braking whereas airbags are deployed during an actual collision. In contrast, use of acoustic signals for automobile pre-crash and crash detection has been somewhat limited even though sounds that occur during such events can offer valuable information. For example, the high-pitch squealing caused by tire skidding can provide advance warning especially if it is caused by an adjacent car. During collision, the acoustic waves traveling along the steel car frame are 17 times faster than the speed of sound in air, which can provide information more promptly than center-mounted acceleration sensors. To fully take advantage of the high-speed acoustic signals, a real-time, wavelet-based, acoustic signal processing method with sub-millisecond precision has been developed, offering distinct advantages in sudden onset detection, temporal localization accuracy, and computational cost over existing time- and frequency-domain methods. Demonstration results on different crash scenarios will be presented, which are indicative of a substantial enhancement of crash detection performance.

8:30

2aSP3. A model of interior cockpit noise for the F-35A. Frank S. Mobley (Airman Systems Directorate, U.S. Air Force, 2610 Seventh St., Bldg. 441, Wright-Patterson AFB, OH 45433, frank.mobley.1@us.af.mil), Alan T. Wall, and Hilary Gallagher (Airman Systems Directorate, U.S. Air Force, Wright-Patterson AFB, OH)

Characterization of environmental noise is vital to protection of hearing. A measurement campaign to understand the nature of the noise in the cockpit of fifth-generation fighter aircraft has been completed. Measurements during multiple missions for each variant of the F-35 aircraft characterized the cockpit acoustic levels during various stages of flight. Additional data collection defined the aircraft state during the acoustic data collection. An initial model of the F-35A represents the A-weighted overall sound pressure level (L_A) within the cockpit by a series of random decision forest models. Sensitivity analysis conducted on these models demonstrates that airspeed and the environmental control system (ECS) flow rate are the two most important parameters determining the interior acoustic levels. However, the measurements cover a subset of the full dynamic envelope of the aircraft. Since machine-learning algorithms are susceptible to greater uncertainty where no acoustic data exists, a curve fit model defines acoustic levels throughout the envelope. This model adds the ECS flow rate to the dynamic pressure calculations and produces a more accurate model of L_A within the cockpit of the F-35A. This research is sponsored, in part, by F-35 Joint Program Office.

8:45

2aSP4. Performance improvement of non-destructive evaluation by steering vector correlation using finite element method. Je-Heon Han (Mech. Eng., Korea Polytechnic Univ., 237. Sangidaehak-ro, Siheung-si, Gyeonggi-do 15073, South Korea, jeep2000@kpu.ac.kr)

In this research, in order to improve the performance of the beamforming algorithm for a nondestructive evaluation, two kinds of steering vector correlation method using the finite element analysis are proposed. First, for non-destructive evaluation applications of reinforced concrete structures, each steering vector is obtained by calculating the responses at the measurement positions for an arbitrary position of a debonded area instead of for an arbitrary position of a source. Then, the reflection effect from the boundaries of concrete samples can be minimized. Second, for the non-destructive evaluations of plates, the finite element analysis is used to investigate the geometric-induced decay model as a function of the distance from a source by applying a curve-fitting method and the obtained steering vector is applied to the MUSIC (MUltiple SIgnal Classification) algorithm to validate its performance.

9:00

2aSP5. Speech, emotion and language: A neuroscientific exploration. Shankha Sanyal (Lang. and Linguist, Jadavpur Univ., Kolkata, West Bengal 700032, India, ssanyal.sanyal2@gmail.com), Archi Banerjee (IIT Kharagpur, Kharagpur, West Bengal, India), Samir Karmakar (Lang. and Linguist, Jadavpur Univ., Kolkata, India), and Dipak Ghosh (Jadavpur Univ., Kolkata, India)

While the emotional state of a person can be manifested in different ways such as facial expressions, gestures, movements and postures, recognition of emotion from speech has gathered much interest over others. This study attempts to understand different factors that influence Speech Emotion Recognition (SER), while taking into account one of the most important parameter—language. We look to classify and compare four basic emotions—anger, happy, sad and neutral from speech segments of four different linguistic groups—Bengali, English, German and Spanish. Robust nonlinear feature like multifractal width was used to develop a language independent emotion classification model. EEG was done on different groups of L1 speaking participants to understand the neuro-cognitive appraisal corresponding to speech segments of different language, i.e., to understand the presence of any speech features that contribute exclusively to emotion recognition of a selected language. The nonlinear, non-stationary EEG time series has been analyzed with latest state of the art multifractal features. The first of its kind study is expected to shed light on how the emotional contents of speech depend on language or solely on the prosodic features of speech such as pitch, loudness, tempo, stress and rhythm.

9:15

2aSP6. Exact FFT-based identification of autoregressive (AR) model. Shigeru Ando (Univ. of Tokyo, 7-3-1 Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, Shigeru_Ando@ipc.i.u-tokyo.ac.jp)

In this study, we extend the weighted integral method (WIM) based on differential equation modeling and finite duration observation [*IEEE Trans. SP* **57**, 9 (2009); *Inverse Problems* **26**, 015011 (2010); and *JASA* **134**(4) (2013)] to a discrete WIM with difference equation (DE) modeling and finite length sampled data sequence, and obtain a novel theory and algorithm for short-time signal analysis and spectral estimation.The discrete WIM is composed of three steps: (1) provide the DE (AR model) with unknown coefficients which is satisfied in finite observation interval. (2) Weighted sum the DE with orthogonal sequences (FFT) to obtain algebraic equations (AEs) among the weighted sums in the interval (discrete Fourier coefficients). A mathematical technique is introduced to maintain circulantness of the time shifts. (3) Solve simultaneously a sufficient number of AEs with least squares criterion to obtain the unknowns exactly when the driving term is absent, or to obtain the ones minimizing the driving power when it is present.This approach will be important both in theoretical aspects and in practical usage for resolution-enhanced time-frequency analysis in multi-harmonics and multi-resonance mixture conditions. We compare the performance with other methods and CRLB, and show several experiments using speech and music signals.

9:30

2aSP7. A learning-based classification of indoor noise type/position in an apartment building. Haesang Yang (Underwater Acoust. Lab., Dept. of Ocean Eng., Seoul National Univ., Bd. 34, Rm. 306, 1, Gwanak-ro, Gwanak-gu, Seoul 151-744, South Korea , coupon3@snu.ac.kr), Hwiyong Choi (Seoul National Univ., Seoul, Gwanak-gu, South Korea), Seungjun Lee, and Woojae Seong (Seoul National Univ., Seoul, South Korea)

This study presents a learning-based method as an indoor noise source type/position classification technique closely related to the interfloor noise problem causing serious conflict between South Korean neighbors. For this study, an indoor noise dataset recorded by a single mobile phone was collected on three floors of an apartment building. Using a convolutional neural networks based classifier, the data containing five source type labels and four position labels is trained and evaluated. In addition, unlabeled data that is not used in the model development is tested for robustness evaluation of our classification system.

9:45

2aSP8. Prediction-error filters for signal and noise separation within ultrasound recordings. Joseph Jennings (Geophys., Stanford Univ., Stanford, CA), Marko Jakovljevic (Radiology, Stanford Univeristy, Stanford, CA), Ettore Biondi (Geophys., Stanford Univ., 397 Panama Mall, Stanford, CA 94305, ettore88@stanford.edu), Jeremy J. Dahl (Radiology, Stanford Univeristy, Palo Alto, CA), and Biondo Biondi (Geophys., Stanford Univ., Stanford, CA)

Ultrasonic image quality can be compromised by several mechanisms such as reverberation, off-axis scattering, and sound-speed inhomogeneities. These mechanisms can cause a loss of coherence of signals across the aperture and clutter in a B-mode image potentially leading to inadequate visualization and misdiagnosis. We develop a processing technique based on prediction-error filters (PEFs) for isolating tissue signal and several types of noise in the individual channel data of ultrasound scans. We first estimate a set of PEFs that allows for prediction of the 2-D spectra of signal and noise in the channel data domain and then use these filters to separate the signal and noise components within the recorded echoes. We use the Van Cittert-Zernike theorem as a priori knowledge of the signal spectrum to develop the signal-PEF, while the noise-PEFs are estimated directly from the channel data. We apply our PEFs to channel data from fullwave simulations of abdominal scans and to data acquired on a lesion phantom through a layer of bovine tissue. The proposed technique successfully separates signal from noise and improves lag-one coherence of the speckle signal from 0.47 to 0.97.

10:00

2aSP9. A new auditory image for social media: Moving towards correlation of spectrographic analysis and interpretation with audience perception. Nguyen Le Thanh Nguyen (Dept. Comput. Sci., Univ. of Miami, 1365 Memorial Dr., Coral Gables, FL 33146, nxn232@miami.edu), Hyunhwan Lee, Joseph Johnson (Marketing Dept., Univ. of Miami, Coral Gables, FL), Mitsunori Ogihara (Dept. Comput. Sci., Univ. of Miami, Coral Gables, FL), Gang Ren (Ctr. for Computational Sci., Univ. of Miami, Coral Gables, FL), and James W. Beauchamp (Dept. Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Spectrogram and other time-frequency analysis methods transfer an audio file into an auditory image. When signal processing-based analysis and interpretation is performed on these auditory images instead of an audio signal, spectrographic analyses can identify interesting patterns that focus on very different aspects of the signal compared to an audio-based analysis. To facilitate an auditory image-based study, a quantitative analysis and interpretation framework is implemented for exploring the spectrographic images in multiple time and frequency scales and for automatically identifying image features that are relevant to human auditory perception. This analysis framework is applied to two social media datasets: (1) soundtracks from video commercials and “hit” music excerpts from social media platforms, and (2) soundtracks from television and film. Analysis results from social media are also compared with audience subjective evaluations to validate the perceptual relevance of the identified spectrographic patterns.

Session 2aUW

Underwater Acoustics, Acoustical Oceanography, and Signal Processing in Acoustics: Comprehensive Nuclear-Test-Ban Treaty International Monitoring System Global Sensor Network: Scientific Aspects and Civil Applications

David L. Bradley, Cochair
University of New Hampshire, 6934 Traveler's Rest Circle, Easton, Maryland 21601

Mario Zampolli, Cochair
IMS/ED, Comprehensive Nuclear-Test-Ban Treaty Organization, PO Box 1250, Vienna 1400, Austria

Georgios Haralabus, Cochair
IMS/ED, Comprehensive Nuclear-Test-Ban Treaty Organization, PO Box 1250, Vienna 1400, Austria

Peter L. Nielsen, Cochair
Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO), Office E0565, VIC, P.O. Box 1200, Vienna 1400, Austria

Chair's Introduction—8:00

Invited Papers

8:05

2aUW1. The Comprehensive Nuclear-Test-Ban Treaty's global International Monitoring System, its core mission and its civil and scientific applications. Mario Zampolli (IMS/ED, Comprehensive Nuclear-Test-Ban Treaty Organization, PO Box 1250, Vienna 1400, Austria, Mario.Zampolli@ctbto.org), Georgios Haralabus (IMS/ED, Comprehensive Nuclear-Test-Ban Treaty Organization, Vienna, Austria), and Peter L. Nielsen (IDC/SA, Comprehensive Nuclear-Test-Ban Treaty Organization, Vienna, Austria)

The Comprehensive Nuclear-Test-Ban Treaty Organization's (CTBTO) International Monitoring System (IMS) comprises 337 stations distributed around the world, recording data 24/7 and sending it in near-real time to the International Data Centre (IDC) in Vienna, where the data are processed to detect signs of nuclear tests. The four monitoring technologies in the IMS are: (i) seismology, (ii) hydroacoustics, (iii) infrasound, and (iv) radionuclide. The present state of completion of the IMS is nearing 90%, with the hydroacoustic network (6 stations with triplets of moored hydrophones suspended in the SOFAR channel and 5 near-shore seismometer stations for T-phase detection) being the first technology to be completely certified as of 2017. In addition to Treaty verification, IMS raw sensor data is available for civil and scientific studies, with hydroacoustic topics encompassing ocean acoustic propagation and tomography, submarine earthquakes and volcanism, ocean soundscapes, marine mammal acoustics and disaster response. This presentation will kick off this ASA Special Session by presenting the current status of the network, with emphasis on the hydroacoustic component and on the hydrophone stations, their sustainment and pertinent technology watch, and by reviewing highlights from data analysis of events for Treaty verification and scientific and civil applications.

8:25

2aUW2. Exploiting ambient noise at hydroacoustic stations for passive ocean sensing. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Conventional acoustic remote sensing techniques typically rely on controlled active sources which can be problematic to deploy and operate over the long term-especially if multiple sources are required to fully illuminate the ocean region of interest- or may not even be available at very low frequencies ($\sim$10 Hz). Conversely, receiver arrays are becoming increasingly autonomous, and capable of long term deployment thus enabling passive acoustics for ocean remote sensing applications by taking advantage of the ubiquitous ocean ambient noise. The archived ambient noise recordings made at the hydroacoustic sessions Comprehensive Nuclear-Test-Ban Treaty (CTBTO) International Monitoring System (IMS), over decades at some locations, provide a unique platform for the scientific community to test this fully passive acoustic approach for ocean remote sensing. This presentation will presents proof of concept of passive ocean remote methods using these hydroacoustic data such as passive acoustic thermometry to estimate deep ocean temperatures variations and internal tides using coherent processing of low-frequency ambient noise. Challenges and opportunities for Ocean basin and global-scale passive ocean sensing will be discussed.

8:45

2aUW3. Some trends and features in deep-ocean noise determined from the CTBTO hydroacoustc stations. Stephen P. Robinson (Ultrasound & Underwater Acoust., NPL, Hampton Rd., Teddington TW11 0LW, United Kingdom, stephen.robinson@npl.co.uk), Peter M. Harris (Data Sci., NPL, Teddington, United Kingdom), Sei-Him Cheong, Lian Wang (Ultrasound & Underwater Acoust., NPL, Teddington, United Kingdom), and Valerie Livina (Data Sci., NPL, Teddington, United Kingdom)

This paper describes the results of applying s statistical method for long term and seasonal trend analysis and uncertainty evaluation of the data from deep-ocean noise data. The analysis method uses a flexible discrete model that incorporates terms that capture seasonal variations in the data together with a moving-average statistical model to describe the serial correlation of residual deviations, with uncertainties validated using bootstrap resampling.The measured data originate from a number of the hydro-acoustic monitoring stations of the CTBTO and span up to a maximum of 15 years. The analysis focuses on the data from Cape Leeuwin Southern Ocean), Wake Island (Pacific Ocean), Ascension Island (Atlantic Ocean), and Diego Garcia (Indian Ocean). The trend analysis is applied to time series representing monthly and daily aggregated statistical levels for five frequency bands to obtain estimates for the change in sound pressure level with associated coverage intervals. The features of the data are described, including the differences observed in the seasonal variation and the long-term trends, with the latter often exhibiting negative gradients. A tentative discussion is initiated of the potential causes of some of the observed trends and fluctuations.

9:05

2aUW4. Hydroacoustic waves traveling along Antarctica: Travel time as a function of the seasons. Läslo Evers (KNMI/TU Delft, PO Box 201, De Bilt 3730 AE, Netherlands, evers@knmi.nl), Pieter Smets, and Shahar Shani-Kadmiel (TU Delft, Delft, Netherlands)

Activities of the Monowai Volcanic centre have been observed as seismic waves recorded at Rarotonga station (RAR), as well as hydroacoustic waves recorded at Ascension Island (H10). Beamforming has been performed on the hydroacoustic data to verify the source and study the effect of performing calculations with a best beam. The lag time that is calculated by cross correlating seismic and hydroacoustic data. With the known source-receiver distance, the sound speed and hence the temperature can be calculated. Finally, the lag time is studied as a function of time in order to observe seasonal changes. In general, travel times between the source and hydrostation H10 do increase with decreasing temperature. In the Southern Hemisphere winter, however, an unexpected decrease of lag time is observed. This might be explained by a different propagation path due to the formation of sea ice.

9:25

2aUW5. Global soundscapes: Parabolic equation modeling and the CTBTO observing system. Kevin D. Heaney (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com) and Anthony I. Eller (Appl. Ocean Sci., Fairfax Station, VA)

The open ocean is not a quiet place, particularly in the Northern Hemisphere. The soundscape of a local environment is defined as detailed understanding of the noise sources observed in time, frequency and space. In this paper we present a set of models and observations of the open ocean sound field utilizing the long-term observations from the Comprehensive Test Ban Treaty Organization (CTBTO) of the United Nations. This system of axial hydrophones spanning the globe in remote locations provides a treasure chest of data for looking at low frequency sound. For frequencies below 125 Hz, half the CTBTO sample rate, the sound field is dominated by local and distant ships, local wind, marine mammals, seismic surveying and seismic events. A basin wide model, with the Parabolic Equation model for propagation, will be presented using Satellite AIS data, surface wind fields and some estimates of fin whale distributions and seismic surveys. The science questions to be addressed are: Can we observe the wind noise or is shipping to dominant? What mechanisms explain wind noise below 50 Hz? Can we use data to move towards a better model for surface shipping source levels? Can we evaluate Fin Whale distributions from regular recordings?

Contributed Papers

9:45

2aUW6. Observation and interpretation of recorded long-range, underwater acoustic signal propagation related to the search of the Argentine submarine ARA San Juan. Peter L. Nielsen (Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO), Office E0565, VIC, P.O. Box 1200, Vienna 1400, Austria, Peter.Nielsen@ctbto.org), Mario Zampolli, Ronan Le Bras, and Georgios Haralabus (Comprehensive Nuclear-Test-Ban Treaty Organization (CTBTO), Vienna, Austria)

In 2017, a signal from an impulse-like event of unknown origin (November) and from a successive controlled experiment in the same area involving the detonation of a depth charge (December), were received at the CTBT IMS hydrophone stations HA10 (Ascension Island, Atlantic Ocean) and HA04 (Crozet Islands, southern Indian Ocean) at ranges greater than 6000 km. The signals propagated through different deep underwater environments along geodesic paths from the sources to the hydrophone stations. The impact of these different environments is clearly observed in the recorded signals, in particular at HA04 as strong time dispersion and low-pass filtering compared to the HA10 data. An interpretation of the signal characteristics is performed by two-dimensional propagation modelling of full time-series including spatially dependent oceanographic database information. Range-dependent bathymetry and typical warm water propagation apply towards HA10, while polar-type conditions with sea-ice extent close to Antarctica are included in the computations at HA04. An acoustic precursor to the main arrival observed at HA04 suggests that the signal partly couples into the ice sheet and partly propagates under the ice sheet. The modelling results broadly agree with observed features and point to the importance of adapting detection and classification algorithms to specific propagation paths.

10:00

2aUW7. On the potential of using sound sources of opportunity recorded on the Comprehensive Nuclear-Test-Ban Treaty hydroacoustic network for historical ocean thermometry. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Ross Heyburn (AWE Blacknest, Reading, United Kingdom)

The globally distributed CTBT hydrophones form an existing network to measure temperature structure of the world oceans. Ocean tomography (or thermometry, asserting the primary driver in sound speed variability is temperature) typically operates by deploying active sources in precisely known locations and projecting signals at precisely known times. Ocean temperature is inferred from the travel time to distributed receivers. Here, we investigate the potential for using sound sources of opportunity within

the nearly 20 year record of CTBT recordings. While some of these historical events have known locations and time, multiple hydroacoustic stations can localize the origin of intense sounds without this information. This study will focus primarily on sounds generated by seismic events detected on the CTBT station Diego Garcia in the central Indian Ocean. Hydroacoustic-based localization is possible through the consideration of multiple refracted and diffracted arrivals caused by the complicated bathymetry near this archipelago. These multiple arrivals are used to form 'virtual' stations to precisely triangulate the sound origin and establish the event time. Comparison of sound source locations to the event epicenters, determined by land-based seismic stations, are discussed in terms of sound generation mechanisms and uncertainties relevant to ocean thermometry.

10:15–10:30 Break

10:30

2aUW8. Comparison of decadal low-frequency ambient noise trends measured in the northern Pacific Ocean: ATOC and CTBTO systems. Rex Andrew (Appl. Phys. Lab., 1013 NE 40th St., Seattle, WA 98105, rex@apl.washington.edu), Kathleen Stafford, and David R. Dall'Osto (Appl. Phys. Lab., Seattle, WA)

Trend estimates from 25 Hz to more than 50 Hz collected from 1994 to 2018 in the north Pacific Ocean are compared. The majority of trends derived from Acoustic Thermometry of Ocean Climate (ATOC) systems over nearly two decades, starting at roughly 1994, suggest a decrease in ambient noise levels of up to 1 dB/year. This is observed on both coastal and deep ocean systems. (Datasets from the remaining systems show either no change or an increase.) Measurements from the Comprehensive Test Ban Treaty Organization (CTBTO) system at Wake Island from 2008 to present provide an overlap from 2008—2013. The CTBTO data streams are essentially continous, and can be averaged and/or aggregated over a variety of time scales. Daily averages show a distinct seasonal cycle. From 2008 to about 2014, the underlying trends are decreasing by as much as 0.55 dB/year, corroborating some of the ATOC observations; from 2014 to the present, the trends are essentially flat. The mechanisms driving these trends appear to be more subtle than the local wind speed or merchant fleet size.

10:45

2aUW9. Multiyear variability of very-low-frequency ambient noise. Anthony I. Eller (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, anthony.eller@appliedoceansciences.com), David L. Bradley (Univ. of New Hampshire, Durham, NH), and Kevin D. Heaney (Appl. Ocean Sci., Fairfax Station, VA)

Several years of continuous low-frequency underwater ambient noise data have been made available for examination by the Comprehensive Nuclear Test Ban Treaty Organization (CTBTO). The data provide ambient noise in the spectral range from 1 to 120 Hz, and studies of the data are in progress at several organizations. Results presented here address the frequency band from 1 to 10 Hz, with special emphasis on discovering evidence of long-term changes in the environment and on quantifying by means of the variance spectrum the contributions of separate noise sources. Long range horizontally refracting propagation and noise source models are used to estimate the contribution of distant events to the observed noise level.

11:00

2aUW10. Subspace detectors for long-term analysis of blue whale vocal activity in the Indian Ocean. Nikita R. Pinto (Dept. of Ocean Eng., Indian Inst. of Technol. Madras, Chennai, Tamil Nadu 600036, India, nikitapinto8@gmail.com) and Tarun K. Chandrayadula (Dept. of Ocean Eng., Indian Inst. of Technol. Madras, Chennai, Tamil Nadu, India)

The Comprehensive Test Ban Treaty Organization (CTBTO) deep water hydrophones around Diego Garcia island (H08) in the central Indian Ocean observe a number of blue whale calls. The hydrophones operational since 2002 are deployed in two groups of three (triads). The first one H08S is to the south, and the second H08N to north of the island. Not much is known about the distribution of whales in the region. The CTBTO observations are hence an opportunity to address this knowledge gap. This work builds subspace detectors to track the whales. The subspaces use the time-frequency signatures of the respective calls. There were however implementation challenges. The call-frequencies drifted across the 16 year dataset. The observed SNRs across hydrophones varied much due to fading. There were also interference issues due to multiple species vocalizing at the same time. This work extends the subspace approach to deal with these issues. The new approach tracks changing frequencies, fuses calls across each triad, and rejects interfering calls. Test results show that the new detectors are relatively robust to false alarm and improve detection rates. The work then applies the detectors to all the H08S and H08N recordings to track whale seasonality and suggest migration patterns.

11:15

2aUW11. Cross ambiguity function mapping of acoustic source tracks using Comprehensive Test Ban Treaty Data. David A. Lechner (Mech. Eng., The Catholic Univ. of America, 9404 Bethany Pl., Washington, DC 20064, 66Lechner@cua.edu), Joseph F. Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC), Shane Guan (Mech. Eng., The Catholic Univ. of America, Silver Spring, MD), and Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This paper presents the adaptation of the Cross Ambiguity Function Mapping (CAF- Mapping) algorithm for use in acoustics. The CAF-Mapping algorithm provides a useful means of generating a geographic visualization of acoustic data. Modifications of the algorithm for use in acoustics are explained, including use of acoustic modeling to generate propagation time differences, nonlinear grid spacing for frequency differences, location seeding using either three sensors or two sensors and two data samples, and frequency normalization. The algorithm allows discrimination between multiple acoustic signals with spatial or velocity vector variance. The presentation compares simulation results for several different types of data, then provides results that track a likely group of pygmy blue whale (Balaenoptera musculus brevicauda) using data from the Comprehensive Test Ban Treaty (CTBT) sensors off the coast of Australia. The author's affiliation with The MITRE Corporation is provided for identification purposes only and is not intended to convey or imply MITRE's concurrence with, or support for, the positions, opinions, or viewpoints expressed by the author. [Approved for Public Release; Distribution Unlimited Case 19-1301. Copyright 2019, The MITRE Corporation.]

11:30

2aUW12. Enabling technology to extend life and utility of hydroacoustic systems. Ryan R. Elliott (Eng., L3Harris MariPro, 1522 Cook Pl., Goleta, CA 93117, ryan.elliott@l3harris.com) and John Reardon (Eng., L3Harris MariPro, Goleta, CA)

The existing infrastructure of the six CTBTO Hydroacoustic System spans remote locations of the world. The first of these systems is reaching its 20 year life and consideration to the maintenance and replacement of the systems is ongoing. This presentation will discuss opportunities to incorporate advancements in active junction box design, facilitating the inclusion of wet-mate underwater fiber optic connections in any future upgrade, replacement, or repair of these systems. Recent analysis of the in-water triplet has shown that these units are likely to last much longer than the 20 year life objective. During any future maintenance or repair, the inclusion of a junction box with wet-mate fiber optic connectors could help to improve the serviceability of the system and could provide opportunity to expand the system to new co-located science research.

Exhibit

An instrument and equipment exhibition will be located in the Ballroom near the registration area and meeting rooms.

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

Monday, 2 December, 5:30 p.m. to 7:00 p.m.: Exhibit Opening Reception including lite snacks and a complimentary drink.

Tuesday, 3 December, 9:00 a.m. to 5:00 p.m.: Exhibit open hours including a.m. and p.m. breaks serving coffee and soft drinks.

Wednesday: 4 December, 9:00 a.m. to 12:00 noon: Exhibit open hours including an a.m. break serving coffee.

Session 2pAA

Architectural Acoustics and Musical Acoustics: Large Music Rehearsing Spaces

Brian Corry, Cochair
Kirkegaard Associates, 7733 Forsyth Boulevard, Suite 1100, St. Louis, Missouri 63105

Kaitlyn Hunt, Cochair
Kirkegaard Associates, 2101 CityWest Blvd, Ste. 204, Houston, Texas 77042

Chair's Introduction—1:00

Contributed Paper

1:05

2pAA1. Data-driven IoT-platform for optimizing variable acoustics in multifunctional performance spaces. Golam Reza Sadeghnia (OKTO Acoust., Gl. Kongevej 11, 4. sal, Copenhagen 1610, Denmark, contact@oktoacoustics.io)

Increasingly more venues are used as multifunctional performance spaces, where a variety of musical genres and spoken word is performed. Churches are turned into public meeting houses, museums are used for concerts and lectures, and concertgoers have an increasing awareness of the acoustical qualities of performance spaces. The problem with multifunctional performance spaces is that different types of performances require different low-frequency acoustical profiles, and changes in the interior of these spaces, audience density and distribution have a significant impact on the acoustic response of a space. Various solutions for variable acoustics have been proposed with predefined fixed settings. However, as of today there is no integrated solution for configuring and validating variable acoustic treatment. We propose a solution based on wireless, fully configurable acoustic panels with variable absorption in the 63 Hz to 8 kHz bands. The mounting method has integrated sensors that communicate with a central model for measuring, calibrating and configuring an assembly of acoustic panels for a certain usage. With several panels mounted on adequate surface areas of a performance space, the system allows parametric control of acoustic treatment to offer optimized variable acoustics. Applications for this type of technology range from educational environments, multipurpose auditoria, and cinemas, to parametric acoustics for mixed reality scenarios.

Invited Papers

1:20

2pAA2. The acoustic design of higher education music rehearsal spaces. Russell A. Cooper (Jaffe Holden Acoust., Inc., Norwalk, CT) and Mark Holden (Jaffe Holden Acoust., Inc., 114A Washington St., Norwalk, CT 06896, mholden@jaffeholden.com)

This paper will update our paper presented in 2007 in New Orleans which analyzed our firm's acoustic design parameters for instrumental, band and choral rehearsal rooms along with measured results. In the subsequent 12 years, we have designed and opened numerous more facilities to add to the databakse of knowledge. Acoustic design parameters such as ideal room size per ensemble type, room dimensions, room height and volume, size and openess of any reflecting clouds, permanent absorption, adjustable acoustic systems, reverberation time and client specific user preferences will be examined. Project examples will be discussed to illustrate the discussion.

1:40

2pAA3. Three music facility makeovers in Arizona. David A. Conant (MCH, Inc., Westlake Village, CA), Arjun K. Shankar (MCH, Inc., 7007 E Gold Dust Ave., Apt. 2089, Scottsdale, AZ 85253, arjunkshankar@gmail.com), and Taylor L. Blaine (MCH, Inc., Scottsdale, AZ)

Considerable prior experience at higher education music campuses in Flagstaff, Chandler and Mesa, Arizona led to our engagement in substantial repurposing and renovation of their rehearsal, recording, practice, and performance facilities. Specific challenges due to existing mechanical system noise, poor sound isolation, and overall room-acoustics issues are discussed as well as their solutions and measured data upon completion.

2:00

2pAA4. Resolving room volume and structural vibration transfer issues at UW Green Bay's Studio Arts Building. John T. Strong (Threshold Acoust., 141 W. Jackson Blvd., Ste. 2080, Chicago, IL 60604, jstrong@thresholdacoustics.com)

The primary rehearsal spaces for the orchestral, band, and choral programs in the Music Department at the University of Wisconsin—Green Bay are hosted at the campus' Studio Arts building, which was constructed in the early 1970's and had not been significantly updated since then. The goal of the building's renovation was to improve room acoustic response for both spaces, improve on the insufficient acoustic isolation between the large instrumental and choral rehearsal rooms, and to update HVAC services. Because of the many years of wear and undocumented alterations to the building, exploratory testing of airborne and structure-borne vibration paths was necessary to determine the contributing means of sound transfer between spaces. Acoustically isolated construction ultimately was utilized to increase separation between rooms as well as to introduce acoustic shaping to the choral rehearsal room, to maximize reverberation and avoid the typical acoustic pitfalls of a roughly cubic room volume. A new reflector array was installed in the large instrumental rehearsal room, as well as targeted absorptive and diffusive treatments. New variable acoustic treatments were designed and installed in both rooms, while noise and vibration control for new HVAC equipment realized suitably low background noise levels for both.

2:20

2pAA5. A more thoughtful approach to the acoustical design of music rehearsal halls. Cameron Goodman (Acoust. Distinctions, Brooklyn, NY, cgoodman@ad-ny.com) and David Kahn (Acoust. Distinctions, Stamford, CT)

With insufficient volume, one must choose between a rehearsal hall that is either too loud or too dead. Increasing room volume helps to reduce this conflict but adds to the cost of construction. Achieving the proper balance between reverberation time, room strength and cost can be challenging. The new 70 733 GSF Music Activities Center for the at Texas A&M University (TAMU) offers four large, purpose-built rehearsal halls for musicians who participate in TAMU bands, choirs, orchestras, and jazz bands, respectively. The design was aimed at creating a balanced acoustic environment with a focus on an acoustically "safe" environment for each of the respective ensembles. The design of these halls was informed by the new Norwegian standard for music facility design, NS 8178, modelling, and close collaboration among design team members.

2:40

2pAA6. A range of large music rehearsal spaces: 3 case studies. Joseph W. Myers (Kirkegaard Assoc., Chicago, IL), Brian Corry (Kirkegaard Assoc., St. Louis, MO), and Kaitlyn Hunt (Kirkegaard Assoc., 2101 CityWest Blvd, Ste. 204, Houston, TX 77042-2830, khunt@kirkegaard.com)

A rehearsal room for a large music ensemble must strike a balance between acoustic support and loudness control. Typically Kirkegaard approaches this by capturing a large volume and treating it with a mix of reflective and absorptive surfaces, plus some movable absorption. The size of the room's volume, the treatment of the reflective and absorptive surfaces, and the location and extent of the adjustable absorption responds to the types of ensembles and music that the room serves. Three examples: At University of Virginia's Hunter Smith Band Building, the main user was the 250-person marching band. This new construction used generous height and deep absorption to control loudness, despite extensive glazing. At University of Southern California's Schoenfeld Symphonic Hall, a former soundstage was adapted into the primary rehearsal room for the Thornton School's orchestra, wind ensemble, and percussion ensembles, with secondary use as a recording space. The renovation had to compensate for limited acoustic volume and directly adjacent mechanical room while transforming the room architecturally. At Oregon Bach Festival's purpose-built Berwick Hall the focus was on early music—rehearsal, instruction, and performance. Our response was a moderate volume with little fixed absorption, creating an unusually reverberant rehearsal room with an extraordinarily beautiful sound.

3:00–3:15 Break

3:15

2pAA7. Sound isolation among music teaching studios with glass curtainwall system. Steven Schlaseman (Jaffe Holden Acoust., Inc., 114A Washington St., Norwalk, CT 06854, sschlaseman@jaffeholden.com)

The acoustic success of music teaching studios is largely dependent on their sound isolation from others. Often these spaces are aligned along exterior walls for natural light and there is an architectural desire to connect them together with lightweight glass curtainwall systems. Sound transfer which flanks through shared curtainwalls and bypasses primary demising construction is a well-documented issue. During the design of the Tianjin Juilliard School in China, Jaffe Holden evaluated options to improve sound isolation for this condition while maintaining the continuous curtainwall aesthetic. This paper discuss the design considerations and the custom sound isolation tests that were performed to analyze and address this issue.

3:35

2pAA8. Vertical campus planning impacts on sound isolation and room acoustics design in New Trier High School's large music rehearsal spaces. Robin S. Glosemeyer Petrone (Threshold Acoust. Co., 53 W Jackson Blvd, Ste. 815, Chicago, IL 60604, robin@thresholdacoustics.com)

By 2015, New Trier High School's student body of 4000 had outgrown its 117-year old campus. With no available land on which to construct new facilities, existing facilities were demolished to allow a vertical campus building to be nestled between the remaining structures. The LEED-Gold certified addition houses the campus' performing arts facilities, a library, cafeteria, applied arts/maker spaces, a culinary lab, interconnected visual arts studios, a technology support bar, STEM laboratories, and flexible classrooms. The performing arts spaces, including five large music rehearsal rooms, are continuously programmed throughout the school day and well into the evening hours, placing high performance requirements on the isolation systems employed in the compact, steel-framed building. The rehearsal rooms employ two distinct sound isolation techniques and room acoustic design approaches as a result of their location within the building and the pre-determined structural spans.

3:55

2pAA9. Tanglewood Learning Institute Linde Center for Music and Learning. Joseph W. Myers (Kirkegaard Assoc., Chicago, IL), Brian Corry (Kirkegaard Assoc., 7733 Forsyth Boulevard, Ste. 1100, St. Louis, MO 63105, bcorry@kirkegaard.com), and Kaitlyn Hunt (Kirkegaard Assoc., Houston, TX)

Since 1940 Tanglewood, the summer home of the Boston Symphony Orchestra in Lenox, Massachusetts, has operated the Tanglewood Music Center (TMC), a summer music academy where students can study with world class musicians. In 2019 Tanglewood added a new program, the Tanglewood Learning Institute (TLI), offering music lovers access to rehearsals, lectures, master classes and recitals. TLI will use a new facility, the Linde Center for Music and Learning, designed by Kirkegaard Associates and William Rawn Associates to complement the nearby 1100-seat Ozawa Concert Hall, designed in the 1990s by the same firms. The Linde Center includes three flat-floor rehearsal rooms, sized at 800 sf, 1,600 sf, and 4000 sf, offering rehearsal space to TMC students during the week, and rehearsal, recital, lecture, and banqueting space for TLI patrons on weekends. The Linde Center is air conditioned, but in keeping with the Tanglewood ethos operable walls and windows allow each of the three rooms to operate with natural ventilation, a challenge for room acoustics and sound isolation. The three spaces share a common architectural language and acoustical approach, but the details of their acoustic designs, AV systems and adjustable absorption respond to the size of the room and the differences in how often they see a particular use.

Contributed Papers

4:15

2pAA10. When a rehearsal space becomes something more—The DiMenna Center for Classical Music, New York, NY. Steve Barbar (E-coustic Systems, 30 Dunbarton Rd., Belmont, MA 02478, steve@lares-lexicon.com) and Anthony Nittoli (Akustiks, Norwalk, CT)

The impetus for developing the Dimenna Center for Classical Music in New York was to create a venue that would enable the Orchestra of St. Luke's—resident orchestra at Carnagie Hall (as well as a select group of additional ensembles)—to rehearse regularly. When the Dimenna Center was envisioned, many of the spaces available for rehearsal in New York were transformed to serve other purposes. Soon after the Dimenna Center opened, it succeeded in its primary mission—more than thirty ensembles are currently resident in the venue. However, the measures taken by the acoustics team in anticipation of providing an optimum rehearsal and recording space for a world class orchestra have also garnered notoriety. Despite its proximity to the subway, the box-in-box construction provides one of the quietest venues of it size for recording in New York. The addition of the E-Coustic System provides acoustic variability that produces optimum acoustical conditions for both rehearsal as well as recording. Recording highlights include the feature film soundtracks for soundtrack for *The Girl On The Train*, *The Greatest Showman*, and *Hail, Caesar!*; Broadway Cast Albums *The King and I*, *Hello, Dolly, and* a Grammy Award-winning album by Room Full of Teeth.

4:30

2pAA11. Correlation between acoustic immersion index and other objective parameters in concert halls. Mojtaba Navvab (Architecture, Univ. of Michigan, 2000 Bonisteel Blvd, Art and Architecture Bldg., Ann Arbor, MI 48109-2069, moji@umich.edu) and Andy Meyer (GFai Tech GmbH, Berlin, Germany)

Concert halls' geometry and volume, impact or shape listeners' room acoustic experience. The current recommended procedure within ISO3382 in calculation or measurement of the room acoustic parameters or current indexes used by designers do not fully represent the listeners' audio experiences. In this paper measured and simulated parameters for room acoustic are analyzed in time and frequency domains for a selected number of well-known halls. Computer simulation and room acoustic measurements using beamforming techniques are used to examine the sound quality and acoustic sense of immersion within these uniquely shaped halls. Halls with the audience facing the stage as compared to the audience's seating surrounding the stage have a noticeable impact on the high-frequency absorption of direct and reflected sounds in each space given the presence of the audience. Correlation between immersion index and other acoustic parameters or indexes such as reverberation time, sound strength, sound texture, listeners' envelopment, early decay and center time, clarity-definition for speech and music, are explored and used to show contributions of acoustical features or key architectural elements within these selected halls. The practical application of these new indexes associated with space geometry for room acoustic design calculations contributes to the high quality of sound experience in concert halls.

4:45

2pAA12. Natural sound (scale) reflecting speaker. Xiaoyu Niu (Inst. of Acoust., Univ. of Chinese Acad. Sci., No. 21, North 4th Ring Rd., Beijing 100190, China, xiaoyuniu-ouc@foxmail.com), Kai Wu (Nanjing Univ., Nanjing, China), Huan Li (Goertek Co., Qingdao, China), Chenguang Li (Peking Univ., Beijing, China), Shang Zhao (Nanjing Univ., Nanjing, China), Gaokun Yu (Ocean Technol., Ocean Univ. of China, Qingdao, Shandong, China), and Shengxue Fu (Ocean Univ. of China, Qingdao, Shandong, China)

In the natural sound music hall, due to rejection of the electronic facilities, engineers need to design an architectural construct to improve the acoustic effect. Rather, it is widely spread to use horn-like construct to improve acoustic energy distributed on the audience. On the other hand, there has not been a suitable nonelectronic device to act as a reflecting speaker. Thus, we propose the concept of Natural sound (Scale) Reflecting Speaker, aiming to control the energy distribution on the stage, so singers could hear their own voice and perform more excellent. In addition, the reflecting speaker consists of multiple coupled Helmholtz resonators. Theoretically, by combining Helmholtz resonances and Bragg scatterings, acoustic negative reflection is achieved at the Bragg blazing points and non-Bragg blazing points. Meanwhile, owing to couplings among these blazing points, broadband, and wide-angle acoustic negative reflection would also be achieved. Finally, we could illustrate the fundamental feasibility of the device via theoretical derivation, computer simulation, and basic experiments. Therefore, it is significant for natural sound music hall to apply to the reflecting speaker in the future.

5:00

2pAA13. Reverberation time slope ratio. Michael W. Fay (none, 7046 Temple Terrace St., San Diego, CA 92119, mfay.gracenote@gmail.com)

T60 Slope Ratio: Symbolically—$T_{60}SR_6$ A proposed standard for condensing six octaves (63 Hz – 2 kHz) of reverberant decay data into a singular-quotient, qualitative score for performance, worship and entertainment facilities. Specifically, a defining metric for scoring and grading the proportional relationship (i.e., ratio) between the longest and shortest of the six T60 values, measured or predicted, and applied to fully enclosed venues employing sound reinforcement systems. In practice, Bass Ratio and Slope Ratio goals are conflicting concepts. Bass Ratio goals and calculations were developed to quantify and support the idea that acoustic instruments need a little extra reverberant support from a room, in the low-frequency range. Slope Ratio goals and calculations support the notion that those same low frequencies do not require extra structural support, but rather need to be managed and well contained. Longer low and very low-frequency T60s are not needed or desirable when an extended-range sound reinforcement system is being used. The $T_{60}SR_6$ thesis is offered to advance and define a room's reverberation goals, and provide a simple numeric scoring scale, and grading vocabulary, from which acoustical design specifications can be initiated and evaluated.

Session 2pAB

Animal Bioacoustics and Acoustical Oceanography: Low-Frequency Sound Production and Passive Acoustic Monitoring II

Jack Butler, Cochair
Marine Physical Lab, Scripps Institution of Oceanography, La Jolla, California 92093

Ana Širović, Cochair
Texas A&M University Galveston, PO Box 1675, Galveston, Texas 77553

Invited Paper

1:30

2pAB1. Exploring kelp forest soundscapes using *ad hoc* frequency-based metrics. Jack Butler (Fish & Wildlife Res. Inst., UCSD/SIO/0205, 9500 Gilman Dr., La Jolla, CA 92093-0205, Jack.Butler@myfwc.com), Jules Jaffe (Scripps Inst. of Oceanogr., San Diego, CA), Ed Parnell (Scripps Inst. of Oceanogr., La Jolla, CA), and Ana Širović (Texas A & M Univ. - Galveston, Galveston, TX)

The kelp forests off the southern California coast are alive with sound, yet their soundscapes remain poorly studied. The kelp forest soundscapes of two sites off La Jolla, CA were surveyed: one within a marine protected area, the other outside of the protected area. Four frequency-based metrics were created to explore how the sounds of the kelp forest vary throughout the day and through the seasons. Two of the metrics tracked frequencies associated with putative fish choruses (60–130 Hz and 300–500 Hz), one metric tracked the long-duration humming of plainfin midshipmen (tonal calls with a fundamental frequency at 85–95 Hz, and harmonics at 175–185 and 265–275 Hz), and the final metric tracked snapping shrimp snaps (2.5–7.5 kHz). The two metrics that tracked the fish choruses exhibited both diel and seasonal periodicity, with strong spectral peaks in the late spring and early summer coinciding with the presence of the choruses. Similar to these two metrics, the metric that tracked the humming of the midshipmen exhibited nightly spectral peaks in the late spring and early summer during their spawning season. The metric that tracked snapping shrimp exhibited strong spectral peaks during dawn and dusk hours yet had little seasonal variability.

Contributed Papers

1:50

2pAB2. Long term PAM to investigate temporal and anthropogenic effects on oyster toadfish, *Opsanus tau* mating vocalizations. Rosalyn Putland (Dept. of Biology, Univ. of Minnesota Duluth, 1035 Kirby Dr., Duluth, MN 55812, rputland@d.umn.edu), Allen F. Mensinger (Dept. of Biology, Univ. of Minnesota Duluth, Duluth, MN), Jacey C. Van Wert (Dept. of Ecology, Evolution and Marine Biology, Univ. of California, Santa Barbara, Santa Barbara, CA), and Alayna Mackiewicz (Dept. of Biology, Univ. of Minnesota Duluth, Duluth, MN)

For the oyster toadfish, *Opsanus tau*, vocal communication and sound detection are critical for reproductive success, however, little is known about how they respond to changes in their acoustic environment. Passive acoustic monitoring was conducted in Eel Pond, MA, USA in the summer months (2017–2019) to investigate vocalization patterns of the resident population and the effect of anthropogenic sound. Male toadfish produce mating vocalizations that are characterized by an initial broadband segment (30–50 ms, 100–1000 Hz) and a longer tonal section (200–650 ms, 100–500 Hz). The pulse repetition rate of the tonal section was significantly related to ambient water temperature during hourly and weekly monitoring. Time difference of arrivals were also used to pinpoint the location of toadfish nests and linked to ambient and anthropogenic sound-maps to understand exposure levels for individual fish. Significantly less vocalizations were detected following exposure to vessel sound (100–12 000 Hz, source level 130 dB re 1 μPa), suggesting individuals changed their vocal behavior in response to anthropogenic activity. Both environmental and the presence of vessel sound influence the acoustic behaviour of toadfish, which could lead to a reduction in communication space, mate attraction and detection.

2:05

2pAB3. Identifying fish sounds of British Columbia with an autonomous audio and video array. Xavier Mouy (School of Earth and Ocean Sci., Univ. of Victoria, 2305–4464 Markham St., Victoria, C V8Z7X8, Canada, Xavier.Mouy@jasco.com), Morgan Black, Kieran Cox, Jessica Qualley, Francis Juanes (Biology, Univ. of Victoria, Victoria, BC, Canada), and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

Among the ~400 marine fish species of British Columbia, only 22 have been reported to be soniferous. However, it is likely due to the lack of examination, as more species are suspected to produce sound. Here we describe how an autonomous audio and video array can identify fish sounds *in situ*. The array is composed of a collapsible PVC frame, an acoustic recorder, six hydrophones, and two custom-made wide-angle autonomous video cameras mounted on the top and side of the array that collect data continuously for up to 10 days. Fish sounds are automatically detected in the acoustic recordings, the time difference of arrivals between pairs of hydrophones is measured by cross correlation, and the 3-D sound-source location and its uncertainty are estimated using linearized inversion. Simulated annealing optimization was used to define the hydrophone configuration that provides

the smallest localization uncertainties. The video recordings are used to assign the species of sound-producing fish localized within the array. The array was deployed at several locations around Vancouver Island and used to define the species, sound characteristics, and source levels of several fish sounds. This new information will help making passive acoustics a viable way to monitor fish in the wild.

2:20

2pAB4. Localization and tracking of fish sounds with a 4-element underwater passive acoustic array. Camille Pagniello (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr. #0205, La Jolla, CA 92093-0205, cpagniello@ucsd.edu) and Gerald L. D'Spain (Scripps Inst. of Oceanogr., San Diego, CA)

Localization and tracking are parameter estimation problems. As such, estimates are evaluated by their statistical distribution, in particular, their bias (equal to the mean of estimated positions minus the true position) and variance (variability in the measured positions about their mean). A variety of methods are used to localize sounds recorded by acoustic arrays. Here, we quantitatively compare the performance of four algorithms: (1) plane-wave-front frequency-domain beamforming; (2) curved-wave-front frequency domain beamforming; (3) time-difference-of-arrivals (TDOAs) using waveform cross-correlation; and (4) TDOAs using spectrogram cross-correlation, for fish sounds generated with a controlled underwater source at various GPS-located positions around a 4-element array. The array consisted of a SoundTrap ST4300 (Ocean Instruments, Auckland, NZ) four-channel acoustic recorder, equipped with four HTI-96-MIN hydrophones (High Tech, Inc., Long Beach, MS). The hydrophones were arranged in a tetrahedral-shaped configuration with a 20-m inter-element spacing. The objective is to optimize the localization and tracking of individual soniferous fish to better understand their small-scale spawning movements and reproductive behavior. Research supported by the Office of Naval Research, the Scripps graduate department, and a Natural Sciences and Engineering Research Council of Canada (NSERC) Postgraduate Scholarship-Doctoral (PGS D-3).

2:35–2:50 Break

2:50

2pAB5. Sound production of the critically endangered totoaba (*Totoaba macdonaldi*): Laying the foundation for conservation of this species through passive acoustic monitoring. Goldie Phillips (Sci-brid Int. Consulting, LLC, 16192 Coastal Hwy., Lewes, DE 19958, g2phillips@ucsd.edu), Gerald L. D'Spain (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), Catalina López-Sagástegui (UC MEXXUSS, Univ. of California, Riverside, Riverside, CA), Daniel Guevara, Miguel Angel Cisneros-Mata (Instituto Nacional de Pesca y Acuacultura Mexico, Guaymas, Mexico), Dennis Rimington (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), David Price (Scripps Inst. of Oceanogr., Univ. of California, San Diego, Point Loma, CA), and Octavio Aburto-Oropeza (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Little is known about the sounds produced by totoaba, a critically endangered fish endemic to the Gulf of California. In a multi-national effort to aid management of this population through passive acoustic monitoring (PAM), we collected simultaneous acoustic and video recordings of totoaba housed at the Centro de Reproducción de Especies Marinas del Estado de Sonora in Kino Bay, Mexico. Measurements were made of both sexes and multiple age classes. Furthermore, individuals were placed in isolation tanks to measure individual call rates and call-to-call variability. Video from these isolation tanks permitted determination of source/receiver range for source level estimation, as well as association of acoustic with physical behaviors, resulting in identification of potential alarm calls. Totoaba produce a variety of calls, ranging from short duration ($<$0.03 s), low-frequency ($<$1 kHz) narrowband pulses to longer, regularly spaced broadband clicks with significant energy over 15 kHz. Call rates and dominant call type depend on both age and sex. Results are used to develop automated pre-processing/detector algorithms for a future PAM system. Quantitative analysis of the performance of these detector algorithms are presented. [Financial support by the Catena Foundation and experiment support by the Centro de Reproducción de Especies Marinas del Estado de Sonora.]

3:05

2pAB6. The effects of tropical storm Debby on soundscapes and marine organisms of the West Florida Shelf (Gulf of Mexico). Anjali D. Boyd (Marine Sci., Eckerd College, 3703 Birmi Dr., Durham, NC 27713, anjaliboyd03@gmail.com), Peter Simard (Environ. Studies, Eckerd College, St. Petersburg, FL), Shannon Gowans (Marine Sci., Eckerd College, St. Petersburg, FL), and David Mann (Loggerhead Instruments, Sarasota, FL)

Tropical cyclones are severe weather systems which can potentially have a large effect on marine ecosystems through direct or indirect effects. In June 2012, Tropical Storm Debby formed in the Gulf of Mexico and had impacted coastal Florida including Tampa Bay. Acoustic recorders were deployed during the storm at a shallow inshore location inside Tampa Bay (Boca 2) and a location offshore in the Gulf of Mexico (Gulf 1). The soundscape before (17–21 June), during (22–26 June) and after (27 June–3 July) Tropical Storm Debby was investigated in two ways: third-octave spectral analysis of root-mean-square sound pressure levels and the identification and quantification of fish sounds in spectrograms. Single-factor ANOVAs indicated a significant increase in ambient noise analyzed in third-octave bands during the storm at both sites ($p < 0.001$), and an overall decrease in fish sound production during the storm at both sites ($p < 0.001$). Several species-specific sound production patterns were also found which correlated with the storm's passage. The changes in ambient noise and biological vocalization was short-lived and returned back to normal within 48 h of the storm. This study is one of three studies to examine the effects of tropical cyclone on marine soundscapes, and the only study to identify sound production to the species level. Furthermore, the results from this study provide important information on the effects of tropical storms on marine communities and the fast rate of recovery after these storms.

3:20

2pAB7. A case study on Bornean gibbons highlights the challenges for incorporating individual identity into passive acoustic monitoring surveys. Dena J. Clink (Bioacoust. Res. Program, Res. Program, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, OR 14850, djc426@cornell.edu) and Holger Klinck (Bioacoust. Res. Program, Cornell Univ., Ithaca, NY)

Passive acoustic monitoring (PAM) has the potential to revolutionize how we study vocal animals, given recent advances in battery life, data storage capabilities and design of recording devices. A major obstacle to the wide-spread implementation of terrestrial PAM programs—particularly with respect to density estimation—is the difficulty in effectively discriminating between calling individuals. Using PAM, identity can be inferred based on caller location or on individually distinct call features. Here, we report the results of a two-month intensive acoustic survey of two gibbon groups in Sabah, Malaysia. We aimed to (1) test a novel acoustic localization method and (2) test how relative distance of recorders to the vocalizing animal, along with recording day and time, influenced our estimates of the call features important for individual identification. We were able to localize calling individuals from within our array with relatively high accuracy, but localization was less accurate at the edges of the array. We found that features estimated from the spectrogram remained relatively stable across recording distances and conditions, whereas Mel-Frequency Cepstral Coefficients were more variable. Our results have important implications for scalability of PAM programs, and based on our results we provide recommendations for incorporating individual identity into PAM programs.

3:35–3:55
Panel Discussion

Session 2pAO

Acoustical Oceanography and Underwater Acoustics: Session in Honor of Michael Buckingham II

Grant B. Deane, Cochair
Marine Physical Lab., Univ. of California, San Diego, 13003 Slack St, La Jolla, California 92093

Simon E. Freeman, Cochair
Scripps Institution of Oceanography, 7038 Old Brentford Road, Alexandria, Virginia 22310

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

Chair's Introduction—1:15

Invited Paper

1:20

2pAO1. Geoacoustic inversion the Buckingham way. N. Ross Chapman (Earth and Ocean Sci., Univ. of Victoria, P.O. Box 3065, Victoria, BC V8P 5C2, Canada, chapman@uvic.ca)

This paper examines the impact of Mike's research on geoacoustic inversion for characterization of sea bed sediments. First is his paper in 1987 that reported the use of vertical coherence of ambient ocean noise for estimating critical angles of sea bed sediment. Apart from the novel use of ambient noise as a sound source, his work opened the idea to make use of information about the ocean bottom contained in spatial phase relationships in vertical hydrophone array data. The practice was quickly adapted and widely applied as matched field inversion. However, the inversions were generally carried out using visco-elastic theory of sound propagation, an approach that is not the most appropriate for applications with porous sediment material. Mike's next contribution addressed this issue in his series of papers starting around 1997 on the grain-shearing and viscous grain-shearing models of sound propagation in porous media. His theory provided a physical basis for the model parameters that are used to describe the interaction of sound with porous sediment media in geoacoustic inversions. Recent examples are shown that indicate how these innovations have become standard practice in geoacoustic inversions.

Contributed Paper

1:35

2pAO2. The analytical approach of Mike Buckingham. Grant B. Deane (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 13003 Slack St., La Jolla, CA 92093-0238, gdeane@ucsd.edu)

Mike Buckingham's contributions to underwater acoustics and acoustical oceanography span a broad range of topics, from field expeditions to measure Arctic ambient noise in the marginal ice zone to a complete model for propagation and attenuation in marine sediments. The hallmark of Mike's work is the formulation of elegant analytical solutions to canonical problems, providing physical insight. I will attempt to illustrate Mike's approach with examples from his work and illustrations of how it has motivated and influenced future approaches.

Invited Papers

1:50

2pAO3. Michael J. Buckingham: Leading the way in ocean ambient noise processing and modeling. Martin Siderius (ECE Dept., Portland State Univ., P.O. Box 751, Portland, OR 97207, siderius@pdx.edu)

Nearly 40 years ago, Michael Buckingham published a theoretical model for ocean ambient noise in shallow water. His pioneering work of the late 1970s and early 1980s advanced not only the methods for understanding the ambient noise field but also the impact on array signal processing. In the late 1980s, Buckingham made a pivotal shift in thinking about ambient noise. He considered ambient noise not as a nuisance factor but as a usable signal. In the late 1980s Buckingham and Jones had the novel idea to take measured ambient noise from a vertical line array and develop an inversion method to determine the seabed critical angle. A few years later,

Buckingham invented a technique known as ambient noise imaging (ANI), or acoustic daylight, which is analogous to conventional photography with daylight as the source of illumination. Buckingham continues to lead the way, and the true significance of his work must include the inspiration his research gives to numerous other scientists, especially those investigating methods to exploit ambient noise as a usable signal. In this presentation, a review of Buckingham's contributions to understanding ambient noise will be described, particularly as they relate to recent methods for ambient noise processing and modeling.

2:05

2pAO4. Some reflections on Buckingham's viscous grain shearing model. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Jan Dettmer (Dept. of Geosci., Univ. of Calgary, Calgary, AB, Canada), and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

Sediment geoacoustic properties are of considerable interest for commercial, industrial, and naval applications. In order to make geoacoustic inferences from acoustic data, the practitioner must choose a sediment acoustics model. The 'Hamilton model' is the most common, which assumes a frequency-independent sound speed, density, and an attenuation varying linearly with frequency, f^1. Buckingham's Viscous Grain Shearing (VGS) model offers a practical alternative with important benefits. For example, since it obeys causality, it provides fundamental bounds to the parameter space that the Hamilton model does not. Furthermore, if the acoustic data are sufficiently informative, the model provides insight into the frequency dependence of sound speed and attenuation. The central physics in VGS are expressed in high-level (material impulse response) functions, which consider both viscous and friction loss mechanisms and lead to attenuation varying between $f^{1/2}$ to f^1 to f^2. Geoacoustic inference from seabed reflection measurements using the VGS model provides insight into the frequency dependence of naturally occurring marine sediments, i.e., admixtures of clay, silt, sand. At present, the VGS model is likely the most general (causal) sediment acoustics model, i.e., can reasonably treat the broadest range of sediment fabrics. [Research supported by the ONR Ocean Acoustics Program.]

2:20

2pAO5. Grain shearing models and relevance beyond sediment acoustics. Sverre Holm (Informatics, Univ. of Oslo, Gaustadalleen 23B, Oslo N 0316, Norway, sverre@ifi.uio.no)

The grain shearing (GS) and viscous grain shearing (VGS) models for saturated, unconsolidated marine sediments fit well to real data. But the models have much further relevance. The inherent convolution is in fact the definition of a non-integer derivative. Therefore, the GS theory is exactly described by the fractional diffusion-wave equation for shear waves and the fractional Kelvin-Voigt wave equation for the compressional mode. Both have been studied extensively in fractional calculus [Pandey and Holm, "Connecting the grain-shearing mechanism of wave propagation in marine sediments to fractional order wave equations," *JASA* (2016)]. The GS theory demonstrates that fractional derivatives arise naturally from physical processes and thus are more than just clever mathematics. These derivatives appear in the relaxation modulus, the stress response to a strain step, of the GS process. The variation in time of the viscosity makes it a time variant non-Newtonian process. But when strain and stress change roles, a surprising result appears. The creep response is the Lomnitz law, an empirically defined logarithmic relationship for such diverse materials as igneous rocks and wood [Pandey and Holm, "Linking the fractional derivative and the Lomnitz…," *Phys. Rev. E* (2016)]. In the VGS model, the power-law response is exponentially tempered. The relaxation modulus is then that of the Cole-Davidson model, known from dielectric materials. This relationship has yet to be explored [Holm, *Waves with Power-Law Attenuation* (Springer, 2019), Chap. 8].

Contributed Paper

2:35

2pAO6. Nonlinear wave propagation in media with attenuation and dispersion modeled using a superposition of relaxation mechanisms. Branch T. Archer (Dept. of Elec. and Comput. Eng., Univ. of Texas at Austin, Austin, TX), John M. Cormack (Walker Dept. of Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX), and Mark F. Hamilton (Walker Dept. of Mech. Eng. and Appl. Res. Labs., Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

Michael Buckingham has long held interest in modeling wave propagation in media with nontraditional attenuation and dispersion, especially unconsolidated marine sediments exhibiting power-law attenuation [Buckingham, *J. Acoust. Soc. Am.* **138**, 2871 (2015)]. The present authors have examined the implications of power-law attenuation and single relaxation mechanisms for modeling nonlinear propagation of compressional and shear waves [Cormack and Hamilton, *Wave Motion* **85**, 13 (2019)]. Reported here are results for media that are modeled using multiple relaxation mechanisms. Below the lowest relaxation frequency, attenuation increases as frequency squared, and above the highest relaxation frequency, attenuation is constant. In between, the superposition may be tailored to approximate power-law attenuation that increases in proportion to frequency raised to an exponent between 0 and 2. Regardless of the source frequency and the number of relaxation mechanisms incorporated, there exists a critical source amplitude above which the mathematical model is incapable of offsetting nonlinear waveform distortion sufficiently to stabilize shock formation beyond the distance where an infinite gradient first appears in the waveform. Results are presented for birelaxing media corresponding to seawater and atmosphere, followed by models based on multiple relaxation mechanisms that approximate power-law attenuation over limited frequency ranges in sediment and tissue.

2:50–3:05 Break

Invited Papers

3:05

2pAO7. On geoacoustic inversions, sediment acoustics, and deep ocean ambient noise. David R. Barclay (Dept. of Oceanogr., Dalhousie Univer., PO Box 15000, Halifax, NS B3H 4R2, Canada, dbarclay@dal.ca)

During the period of 2005–2011, three topics in acoustical oceanography were actively investigated in the Buckingham Lab which combined field, lab, and theoretical work. Geoacoustic inversions in the low frequency (<500 Hz) band were carried out using the noise generated by a light aircraft off the coast of La Jolla, California, and during the MAKAI experiment in Kauai, Hawaii. The inversion relied on a comparison of data to an analytical model of a moving airborne source over a two-layer shallow water waveguide. In an effort to investigate the link between grain roughness and porosity to support the development of a theory of acoustic propagation in sediments, a statistical model for the two-dimensional shape of sand grains was developed from the digitized outlines of grains from a dozen locations, including deserts, beaches, and seabeds. Lastly, a family of autonomous free-falling ambient noise profilers known as Deep Sound, were designed, manufactured, and deployed, returning measurements of ambient noise from below the critical (or reciprocal) depth in the deep ocean. Field experiments were carried out in the San Diego trough, the Philippine Sea, the Tonga Trench, and the Sirena Deep and Challenger Deep in the Mariana Trench.

3:20

2pAO8. Ambient noise in coral reef environments. Simon E. Freeman (Naval Undersea Warfare Ctr., 7038 Old Brentford Rd., Alexandria, VA 22310, simon.freeman@gmail.com) and Lauren Freeman (Naval Undersea Warfare Ctr., Newport, RI)

Ambient noise in very shallow water is often dominated by sound from biological sources. In addition to their economic and aesthetic value, coral reef environments are of interest acoustically due to their persistently loud biological ambient noise levels, relatively high biomass, diversity of soniferous organisms, and their relatively unique spatial nature in which the receiver is often surrounded by large numbers of transient sources, each emitting relevant information. Stemming from work in the Buckingham lab, our investigation of these sounds and their connection with biological processes continues through both directional and point receivers. This talk will discuss how information contained in the transient aspects of these sound fields can reveal the nature of the sources and physical mechanisms behind the generation of their sounds, spatial distribution of benthic communities, dynamic interactions between organisms and changes in the environment arising from human impacts such as the removal of herbivorous fishes.

3:35

2pAO9. Estimating low-frequency (<100 Hz) sound speed in fine to very-fine sand sediment from the horizontal coherence of the head wave excited by a light helicopter. Dieter A. Bevans (Marine Physical Lab., Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093-0238, dbevans@ucsd.edu) and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

A series of shallow-water acoustic experiments has been conducted off La Jolla, California using a Robinson R44 helicopter as a low-frequency (≈13–2500 Hz) sound source. The sound speed of the sand sediment was recovered from the horizontal coherence of the head wave excited in the water column by the helicopter. Previously, on 14 December 2016, with a horizontal aperture of 3 m, this technique returned the sound speed of the sediment, from 800–2300 Hz, as 1682 ± 16 m/s, consistent with the known sediment type. An additional experiment was conducted on 6 February with larger horizontal apertures of 9, 12, and 15 m. The hydrophones, situated 0.5 m above the seabed, received the head-wave signals, allowing the coherence function to be formed over the bandwidth of the airborne source. The sediment sound speed was recovered by matching the zero crossings of the measured coherence function to those predicted by the theory of head-wave generation in shallow water. The dispersion in the sediment sound speed was estimated over a frequency range extending between 27 Hz, the lowest zero crossing of the coherence function, and 2.5 kHz, the bandwidth of the source. [Research supported by ONR, SMART(DOD), NAVAIR and SIO.]

Contributed Papers

3:50

2pAO10. The effect of submarine canyon bathymetry on range estimation using cross-correlated ship noise field. Najeem Shajahan (Oceanpgraphy, Dalhousie Univ., 1355 Oxford St., PO Box 15000, Halifax, NS B3H 4R2, Canada, nj210471@dal.ca) and David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

Ambient noise data collected on a pair of omni-directional vertically separated hydrophones at the head of Alvin Canyon, a shelf-break submarine canyon, is used to estimate the range and bearing of a passing vessel. Comparison of the vertical coherence data against a Pekeris normal-mode model may be used to provide the range of the ship along bearing lines that do not cross the canyon's rapidly changing bathymetry. To investigate the effect of bathymetry on the range estimate, a reciprocal three-dimensional Parabolic Equation (3-D PE) model is used to generate a map of the vertical coherence field for all possible ship positions over the domain of a Gaussian canyon, demonstrating that horizontal reflection and refraction lead to focusing of ship noise along the canyon axis. The same method is used to obtain vessel range and bearing information from the data recorded at Alvin Canyon. The vessel bearing relative to the pair of vertically spaced omni-directional receivers is obtained by exploiting the canyon bathymetry.

4:05

2pAO11. Moving ship tomography in the Santa Barbara Channel Experiment 2016. Kay L. Gemba (MPL/SIO, UCSD, Univ. of California, San Diego, 8820 Shellback Way, Spiess Hall, Rm. 446, La Jolla, CA 92037, gemba@ucsd.edu), Heriberto J. Vazquez, Jit Sarkar, Bruce Cornuelle, William S. Hodgkiss, and William A. Kuperman (MPL/SIO, UCSD, La Jolla, CA)

Ocean acoustic travel-time-tomography attempts to map heterogeneities such as ocean temperature with mesoscale resolution. The Santa Barbara Channel Experiment in 2016 (SBCEx16) contained both controlled and uncontrolled (i.e., ships of opportunity) sources, in addition to environmental data collected by a variety of sensors. The path to developing a methodology for utilizing noise from ships of opportunity for oceanographic imaging involves benchmarking our processing with a controlled source-tow. This latter moving source tomography, itself, is a procedure that still remains to be quantitatively validated and documented. We present results from the source-tow tomographic component of SBCEx16 with the added objective of understanding the requirements for minimizing the uncertainties associated with uncontrolled, random sources of opportunity.

Invited Paper

4:20

2pAO12. Michael Buckingham—Cosmologist. James Lynch (Woods Hole Oceanographic, MS # 11, Woods Hole Oceanographic, Woods Hole, MA 02543, jlynch@whoi.edu)

Before Mike Buckingham became and ocean acoustician, he had a rather different career path—that of a cosmologist trying to detect gravitational waves. In the late 1960s, physicist Joseph Weber published a paper in which he claimed to have measured Einstein's gravitational waves. Efforts quickly ensued to either confirm or disprove Weber's stunning claim—and Mike Buckingham was in the thick of these efforts. The story of Mike's involvement traces a very interesting episode in the history of gravitational wave research. It also involves a fair bit of acoustics, as that was the technique used at that time (and is still pursued to some extent today). Perhaps most happily, this story explains how and why Mike Buckingham became an ocean acoustician, which is the ASA's gain (though cosmology's loss.)

Contributed Paper

4:35

2pAO13. Personal reminiscences of a career in acoustical oceanography. Michael J. Buckingham (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

Over recent years, ambient noise in the ocean has come to be recognized as a signal in its own right, containing a wealth of information about the ocean and ocean processes. Ambient noise has thus become a central pillar of acoustical oceanography. Since the attributes of ambient noise are based on straightforward physical principles, the noise field may be inverted to recover, for example, the sound speed in the seabed. Some selective examples of ambient noise oceanography will be reviewed, based on personal experiences gained during a career that includes research in the Arctic, the North Atlantic, the deep ocean trenches, and the shallow, continental shelf waters of the English Channel and the Pacific Ocean, off the coast of California. Much of this work was accomplished with the talented assistance of many students and colleagues; and it was made possible through the consistent support of our sponsors. It is a pleasure to acknowledge that, throughout, the Acoustical Society has provided a cordial environment where acoustical oceanographers could meet and are always made welcome. [Research supported by ONR.]

TUESDAY AFTERNOON, 3 DECEMBER 2019 HANOVER, 1:15 P.M. TO 4:00 P.M.

Session 2pBAa

Biomedical Acoustics and Signal Processing in Acoustics: High Frame Rate Ultrasound Imaging: Technical Developments and Clinical Applications II

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

Alessandro Ramalli, Cochair
Cardiovascular Imaging on Dynamics, KU Leuven, Leuven 3000, Belgium

Invited Papers

1:15

2pBAa1. GPU technology in the high frame rate ultrasound imaging era. Billy Y. Yiu, Adrian J. Chee (Univ. of Waterloo, Waterloo, ON, Canada), and Alfred C. Yu (Univ. of Waterloo, EIT 4125, Waterloo, ON N2L 3G1, Canada, alfred.yu@uwaterloo.ca)

As a rapidly maturing class of many-core parallel computing hardware, graphical processing unit (GPU) has enjoyed a surge of interest in the medical ultrasound community in the past decade. In particular, the software-level programmability of GPUs has significantly lowered the entry barrier for ultrasound imaging researchers (who might not be parallel computing specialists) to pursue fast realization of novel imaging algorithms that have known theoretical potential but have yet to demonstrate their real-time feasibility. This presentation shall highlight how GPUs have emerged as a new computing workhorse in realizing various ultrasound imaging innovations. Of particular note is the enabling role that GPUs have played in fostering practical pursuit of high-frame-rate ultrasound (HiFRUS) imaging innovations that are based on direct processing of pre-beamformed radiofrequency (RF) data acquired from individual array elements. Using the state-of-art GPU technology, it is readily possible to achieve >1,000 fps HiFRUS beamforming throughput. GPU can also play a pivotal role in facilitating fast realization of computationally intensive HiFRUS algorithms, such as adaptive beamforming, color-encoded flow speckle imaging, and eigen-processing. The real-time performance of these GPU computing kernels will be discussed, and their practical implementation on software-oriented open-platform research systems will be presented.

1:35

2pBAa2. The contribution of research platforms to the pathway from off-line to real-time high-frame-rate imaging. Piero Tortoli (Information Eng., Università di Firenze, via Santa marta 3, Firenze 50139, Italy, piero.tortoli@unifi.it) and Enrico Boni (Information Eng., Università di Firenze, Firenze, Italy)

Ultrasound research platforms are characterized by a great flexibility of parameters that are usually fixed on clinical scanners. They allow the transmission of arbitrary waveforms in arbitrary scan sequences, and to test novel methods by processing off-line the raw radio-frequency channel data received after each transmission (TX) event. In particular, the availability of research platforms has boosted the development of high-frame-rate (HFR) imaging methods. These are typically based on the TX of multi-line focused beams, plane waves or diverging waves, which can be optimally produced by linear power amplifiers. The echoes received after each TX event should thus be beamformed along multiple lines in parallel, which is feasible, on line, only when high processing power is available. In this talk, the characteristics of "hardware-based" research platforms, like the ULA-OP 256, are discussed. The presence of powerful FPGA devices, on board such systems, allows implementing efficient parallel beamforming methods, which is the key-requirement for HFR imaging. Sample results obtained in real-time at rates of hundreds frames/s will be presented for both multi-line and plane wave compounding scan sequences. Finally, the main challenges to be faced to extend such modalities to volumetric (3-D) imaging will be discussed.

1:55

2pBAa3. Filtered delay multiply and sum beamforming in high frame-rate ultrasound imaging. Giulia Matrone (Dept. of Elec., Comput. and Biomedical Eng., Univ. of Pavia, Via Ferrata 5, Pavia 27100, Italy, giulia.matrone@unipv.it) and Alessandro Ramalli (Dept. of Cardiovascular Sci., KU Leuven, Leuven, Belgium)

Filtered Delay Multiply and Sum (F-DMAS) is an ultrasound beamforming technique derived from microwave imaging, which exploits the spatial autocorrelation of the receive aperture to improve the quality of images. Being based on the spatial coherence of backscattered echoes, it reconstructs a pulse-echo beam with lower sidelobes and a narrower main lobe as compared to standard Delay and Sum (DAS). Thanks to this, F-DMAS can be effectively employed in high frame-rate (HFR) ultrasound imaging, where higher acquisition rates come together with more artifacts, lower contrast and worse spatial resolution. This presentation will provide an overview of the most recent applications of F-DMAS in HFR imaging, e.g., with multi-line transmission and plane-wave techniques. Results of simulations, phantom experiments and *in vivo* scans, particularly of cardiac imaging, show that F-DMAS represents an effective solution to improve image quality, suppress artifacts and increase the frame-rate at the same time. Moreover, two alternative methods, developed starting from the F-DMAS formulation, will be shown as they further enhance the image quality in HFR imaging.

2:15

2pBAa4. Short-lag spatial coherence imaging with multi-line transmission to improve needle visibility in ultrasound images. Giulia Matrone (Dept. of Elec., Comput. and Biomedical Eng., Univ. of Pavia, Via Ferrata 5, Pavia 27100, Italy, giulia.matrone@unipv.it), Muyinatu A. Lediju Bell (Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD), Eduardo Gonzalez (Dept. of Biomedical Eng., Johns Hopkins Univ., Baltimore, MD), and Alessandro Ramalli (Dept. of Cardiovascular Imaging and Dynam., KU Leuven, Leuven, Belgium)

Ultrasound Short Lag Spatial Coherence (SLSC) imaging exploits the spatial coherence (SC) of backscattered echoes to suppress acoustic clutter and to significantly improve image contrast, particularly in difficult-to-image patients. Multi-line transmission (MLT) imaging simultaneously transmits multiple beams to increase frame rates. We hypothesize that the combination of SLSC and MLT has the potential to enhance detectability of coherent structures within a tissue background. Sector images of a 14G biopsy needle inserted within *ex vivo* bovine meat were acquired after implementing single-line transmission (SLT) and MLT with 4, 6, 8 and 16 simultaneous beams. The SLSC algorithm was applied to the acquired data. Results show that MLT SLSC images have a visibly darker background than SLT SLSC images, which is caused by the rapid decorrelation of the SC at short lags with MLT. This decorrelation is caused by inter-beam crosstalk interferences and can be leveraged to enhance the coherence of highly reflective targets within a tissue background, causing an increase in needle contrast from about 25 dB with SLT to ~44 dB with 16-MLT. These results support our hypothesis that the decorrelation caused by crosstalk improves visualization of highly coherent targets within tissue, with promising applications to biopsy needle localization.

2:30

2pBAa5. High frame rate color flow echocardiography: A comparison of different imaging approaches. Alessandro Ramalli (Cardiovascular Imaging and Dynam., KU Leuven, UZ Herestraat 49 - Box 7003, Leuven 3000, Belgium, alessandro.ramalli@unifi.it), Alfonso Rodriguez-Molares, Jorgen Avdal (Dept. of Circulation and Medical Imaging, NTNU, Trondheim, Norway), Jan D'hooge (Cardiovascular Imaging and Dynam., KU Leuven, Leuven, Belgium), and Lasse Lovstakken (Dept. of Circulation and Medical Imaging, NTNU, Trondheim, Norway)

Quantitative measurements by color flow imaging (CFI) are hindered by both low temporal resolution and a limited field of view. These limitations can be overcome by recent high frame rate (HFR) imaging techniques. However, as these HFR techniques reduce image quality and penetration depth to a different degree, the aim of this study was to compare the impact of HFR techniques on CFI. Hereto, phantom and *in-vivo* images were recorded using a cardiac phased array (P4-2v) connected to a Vantage 256 system (Verasonics) in order to scan a 90 deg wide, 12 cm deep sector. Different HFR scan sequences were tested, including multi-line transmission with 4 simultaneously transmitted beams (4MLT), diverging waves with 2 different opening angles, $2\varphi = 20$ deg (DW20) and 90 deg (DW90), as well as single-line transmission (SLT) as a benchmark. For a fair comparison and to evaluate a clinical applicability, the acoustic output was equalized for all sequences, for which the heating of the probe was the main restriction. Results show that HFR techniques spread artifacts on larger areas compared to SLT (>+50%), but the velocity estimates are still comparable (relative error < 6%). 4MLT and DW20 achieved similar performance and enable wide-angle CFI at HFRs (78 Hz in continuous acquisition over 90 deg). On the other hand, DW90 boosts the frame rate up to 625 Hz but with a reduction of the penetration depth (up to −5/6 cm).

2:45

2pBAa6. Filter optimization for orthogonal frequency division multiplexing combined with multi line transmission for ultrafast ultrasound imaging. Yue Song (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Trento, Italy), Alessandro Ramalli (Dept. of Cardiovascular Sci., KU Leuven, Leuven, Belgium), Enrico Boni (Dept. of Information Eng., Univ. of Florence, Firenze, Italy), and Libertario Demi (Dept. of Information Eng. and Comput. Sci., Univ. of Trento, Via Sommarive, 9, Trento 38123, Italy, libertario.demi@unitn.it)

Multi-line transmission (MLT) and orthogonal frequency division multiplexing (OFDM) are two types of approaches which have been individually applied to increase ultrasound data acquisition rate. Both consist in the simultaneous transmission of several beams to reconstruct multiple image lines at the same time, which, however, implies interbeam cross-talk that deteriorates image quality. In particular, as a consequence of interbeam cross-talk, imaging artifacts are generated. While MLT essentially relies on beams spatial-separations to limit interbeam cross-talk, OFDM mitigates interbeam-cross-talk by assigning to each beam a fraction of the transducer bandwidth. Recently the combination of the two techniques has been tested and shown effective to further boost the data acquisition rate [doi:10.1109/ULTSYM.2018.8580107]. In this work, results from a filter optimization study are reported and discussed. Data as obtained from wire targets and tissue mimicking phantoms have been analysed to asses the impact of filter type, order and bandwidth on key imaging features, i.e., axial resolution, interbeam cross-talk, and artifacts area. The ULA-OP 256 system equipped with a 2.4 MHZ phased array was employed to collect the data.

3:00–3:15 Break

3:15

2pBAa7. Multi-pulse, plane-wave acoustic contrast agent imaging with a 16 MHz linear array. Jeffrey A. Ketterling (Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., New York, NY 10006, jkettermeister@gmail.com) and Ronald H. Silverman (Dept. of Ophthalmology, Columbia Univ. Irving Medical Ctr., New York, NY)

Plane-wave imaging allows high-frame rates, and with high-frequency (HF) ultrasound, fine-spatial and temporal resolution. HF, plane-wave multi-pulse (MP) imaging methods of acoustic contrast agents (UCAs) are still an unexplored area for small-animal and small-parts imaging. The MP methods of pulse inversion (PI, + 1 and -1 pair) and amplitude modulation (AM, + 1 and + ½ pair) were compared to standard plane-wave compound imaging for a flow of the UCA Definity in a linear-channel, wall-less flow phantom using a 16-MHz linear array. Transmissions were made at -10, 1 and 10 deg. Definity was diluted in water to 2000:1 and a flow was established through the phantom. Beamformed data were processed to find RMS values in a flow and background region in order to calculate contrast-to-noise ratio (CTR). The MP methods resulted in an 8 dB improvement in CTR relative to standard plane-wave imaging. The absolute intensity of the flow and background regions dropped for PI and AM relative to standard plane-wave imaging. At all acoustic drive pressures, PI and AM CTRs were statistically different than with standard imaging ($p < 10^{-4}$). PI and AM CTRs were statistically different above 1 MPa ($p < 10^{-4}$) where PI had a 3 dB higher CTR.

3:30

2pBAa8. A convolutional neural network approach to preserve image quality for sparse array data. Di Xiao (Elec. and Comput. Eng., Univ. of Waterloo, 250 Laurelwood Dr., Waterloo, ON N2V 2R4, Canada, di.xiao@uwaterloo.ca), Billy Y. Yiu, Adrian J. Chee, and Alfred C. Yu (Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

High-frame-rate ultrasound (HiFRUS) has seen recent interest in resolving highly dynamic spatiotemporal events but transferring the large amount of data generated from the probe remains a hurdle for real-time imaging. One method to lessen the data traffic while preserving the field of view is to reduce the channel count, but this can lead to image quality degradation and the appearance of grating lobe artifacts. In this work, we present a convolutional neural network (CNN) based framework that uses a sparse array (half the channel count) and infers the remaining channels to mimic a fully-populated array. On unfocused transmissions, our results show that on a beamformed image of a multiple point target phantom, grating lobe artifacts are reduced from over 8dB (sparse array) to less than 1 dB (CNN interpolated) when compared to an image beamformed using the full array. Additionally, reconstructions from CNN generated data demonstrated improvement (10 dB) in carotid echolucent flow regions *in vivo*. Our work demonstrates that, using a deep learning approach to channel-domain radiofrequency data interpolation, the required physical channel count on an array and the corresponding data transfer bandwidth can both be reduced without significant image quality trade-off.

3:45

2pBAa9. Frequency domain beamforming of ultrasound signals from inhomogeneous media using range Doppler method. Marko Jakovljevic (Radiology, Stanford Univ., Stanford, CA 94305, marko.jakovljevic@stanford.edu), Roger Michaelides, Ettore Biondi, Howard Zebker (Geophys., Stanford Univ., Palo Alto, CA), and Jeremy J. Dahl (Radiology, Stanford Univ., Palo Alto, CA)

Frequency domain beamforming methods, such as Stolt migration, yield efficient and fast processing of ultrasound data. Stolt migration, by migrating the individual element data in a 2-D frequency domain, improves processing speed by up to two orders of magnitude compared to conventional delay-and-sum (DAS) beamforming. However, as commonly implemented, it relies on the assumption that the speed-of-sound is constant, and is unable to be applied with advanced beamforming techniques such as phase aberration correction. Here, we demonstrate another approach to frequency domain beamforming, using the range-Doppler method developed for synthetic aperture radar imaging. The range-Doppler method is similar in speed to Stolt migration, but because it involves only a 1-D Fourier transform in array dimension, it can account for speed-of-sound inhomogeneities. We demonstrate our method with simulated channel signals of point targets and diffuse scatterers, and find similar image quality (e.g., mainlobe width, sidelobe levels, and contrast-to-noise ratio) to DAS beamforming in noise-free conditions. We further analyze signals acquired through a near-field phase-screen and from multi-layered media by modifying Doppler processing to correct for phase aberration and to account for the local sound-speed differences, respectively.

TUESDAY AFTERNOON, 3 DECEMBER 2019 GARDEN, 1:15 P.M. TO 4:50 P.M.

Session 2pBAb

Biomedical Acoustics: Application of Quantitative Ultrasound *in vivo* in Humans II

Jonathan Mamou, Cochair
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Michael Oelze, Cochair
UIUC, 405 N Mathews, Urbana, Illinois 61801

Invited Papers

1:15

2pBAb1. Passive elastography: Extracting mechanical properties of soft tissues from natural elastic waves. Stefan Catheline (INSERM U1032, 151 cours albert thomas, Lyon 69003, France, stefan.catheline@inserm.fr), Bruno Giammarinaro (INSERM U1032, Lyon Cedex 03, France), Ali Zorgani (INSERM U1032, Bron, France), Chadi Zemzemi, Rémi Souchon, and Olivier Rouvière (INSERM U1032, Lyon, France)

Clinical devices that give elastography imaging from a shear wave speed measurement within tissues uses controlled sources for elastic wave generation. It can be a mechanical vibrator for transient elastography and magnetic resonance elastography (MRE) or ultrasound radiation pressure for ARFI and shear wave elastography technique. Generally speaking, the main drawback of all these methods is the drop of efficiency for deep regions and obese patient. Delivering robust elastic waves in depth is an issue. We have proposed in the past a solution that avoid the problem of shear wave penetration using active sources. It consists in using natural shear waves that exist everywhere in living tissues. These natural shear wave are created thanks to heart beating, to pulse wave within arteries or to muscle activities. Since a few years now, we have been tracking natural shear waves in different soft tissues of the human body and we have not yet been disappointed. Using uncontrolled shear wave needs a change in the group velocity measurement paradigm. This approach is the so-called noise correlation technique developed in the field of seismology.

1:35

2pBAb2. Ultrasound elastography assessment of increased carotid artery wall stiffness in human immunodeficiency virus infected patients. Guy Cloutier (Univ. of Montreal Hospital Res. Ctr., 2099 Alexandre de Sève, Montreal, PQ H2L 2W5, Canada, guy.cloutier@umontreal.ca), Marie-Hélène Roy Cardinal, Carl-Chartrand Lefebvre, Gilles Soulez, Cécile Tremblay, and Madeleine Durand (Univ. of Montreal Hospital Res. Ctr., Montreal, PQ, Canada)

Prevalence of cardiovascular diseases is higher in HIV subjects when compared to the general population. The chronic immune stimulation and low grade inflammation are possible causes. Carotid artery strain and motion are proposed as markers of premature atherosclerosis. Seventy-four HIV infected and 75 age-matched control subjects were recruited from a prospective, controlled cohort study (mean age 56 years ± 8; 128 men). Elastography applied to longitudinal ultrasound images of common and internal carotid arteries quantified the cumulated axial strain, cumulated shear strain, cumulated axial translation, and cumulated lateral translation. Presence of plaque was also assessed. Association between elastography biomarkers and HIV status were evaluated with Mann–Whitney tests and multivariate linear regression models. A higher occurrence of carotid plaques was found in HIV infected individuals ($p = 0.011$). Lower cumulated lateral translations were found in HIV infected subjects on both common and internal carotid arteries ($p = 0.037$ and $p = 0.026$, respectively). These observations remained significant when considering multivariable models including common cardiovascular risk factors and clinical characteristics ($p < 0.05$). Lower cumulated axial strains were also observed in internal carotid arteries when considering both multivariate models ($p < 0.05$). Lower translation and strain of the carotid artery wall in HIV infected individuals, which indicate increased vessel wall stiffness, could be new biomarkers of premature atherosclerosis.

1:55

2pBAb3. Using multiparametric ultrasound-based elastographic characterization for evaluation of renal transplants. Matthew W. Urban (Dept. of Radiology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu), Luiz Vasconcelos (Biomedical Informatics and Computational Biology, Univ. of Minnesota, Rochester, MN), and Piotr Kijanka (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Renal transplant is a long-term therapy for patients in end-stage renal disease. Renal transplant health is monitored using laboratory tests and renal biopsy. However, renal biopsy is invasive and has issues with sampling errors. Ultrasound-based shear wave elastography (SWE) is being evaluated in renal transplants to identify patients with early stages of interstitial fibrosis and inflammation. We are exploring the use of new image reconstruction approaches such as local phase velocity imaging (LPVI) to evaluate viscoelasticity in a cohort of 23 renal transplants. Additionally, from point SWE measurements we have compiled results from 97 patients and have used various machine learning approaches to differentiate patients without disease such as interstitial fibrosis, inflammation, or interstitial fibrosis with tubular atrophy (IFTA). The LPVI method provides two-dimensional maps of phase velocity and images of viscoelastic parameters after fitting to a rheological model. The machine learning approaches combine multiple elastographic parameters to provide differentiation with high sensitivity, specificity, and accuracy (∼ 85%–90%). These approaches could have substantial benefit for patients for identifying early disease in renal transplants that my precede changes in laboratory test results.

2:15

2pBAb4. Ultrasonic evaluation of bone health in patients. 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)

Noninvasive and quantitative methods for evaluation of bone health are of interest in clinical practice. Two areas of particular interest include evaluation of bone fracture healing and assessment of osteoporosis. Determination of bone healing time impacts patient health and the associated economic factors. Furthermore, assessment of bone quality in patients at risk for osteoporosis/osteopenia can benefit a large number of patients in the upper age group. Here, we present a quantitative method for the evaluation of bone integrity and quality. This method is based on analyzing the acoustic response of bone to stimulation provided by the acoustic radiation force of ultrasound. Longitudinal studies for bone healing assessment were conducted on a group of patients with clavicle fracture. The contralateral intact bone was used as a control for each patient. Bone healing assessments were performed using mode decomposition. It was found that the healing process could be detected in 80% of fractured bones. A separate study was conducted to differentiate osteoporotic/osteopenic from normal bones. A total of 27 adult volunteers in two groups of healthy and those with osteopenia/osteoporosis were tested. Results of the test on tibia showed that osteoporosis/osteopenia bones could be differentiated from the healthy ones ($<2 \times 10^{-5}$).

Contributed Papers

2:35

2pBAb5. Relationships among quantitative ultrasound-assessed hepatic health status, gut microbiota, and bile acids among adults with overweight or obesity. Sharon V. Thompson (Div. of Nutritional Sci., Univ. of Illinois, 905 S. Goodwin, Urbana, IL 61801, svthomp2@illinois.edu), Aiguo Han (Univ. of Illinois, Urbana, IL), Caitlyn G. Edwards, John W. Erdman (Div. of Nutritional Sci., Univ. of Illinois, Urbana, IL), W. D. O'Brien, Jr. (Univ. of Illinois, Urbana, IL), Naiman A. Khan (Dept. of Kinesiology and Community Health, Univ. of Illinois, Urbana, IL), and Hannah D. Holscher (Dept. of Food Sci. and Human Nutrition, Univ. of Illinois, Urbana, IL)

Nonalcoholic fatty liver disease (NAFLD) is a progressive condition characterized by hepatic fat fraction (FF) ≥ 5%. The gut-microbiota-liver axis has been implicated in NAFLD development. Aim: to assess microbial taxa, bile acid concentrations, and FF among adults with overweight or obesity; cross-sectional study was conducted among adults (n = 118; 25–45 yr, BMI ≥ 25.0 kg/m^2). NAFLD was assessed using quantitative ultrasound FF estimates based on attenuation and backscattered coefficients. Microbial analyses were conducted using 16S rRNA sequencing (V4 region) and QIIME2 version 2019.4. Fecal bile acid concentrations were measured using LC-MS/MS. Bivariate correlations assessed relationships between FF and microbes or bile acids. General linear model univariate tests identified microbial taxa differences by presence or absence of NAFLD. Hepatic FF was positively correlated with Proteobacteria (r = 0.21, p = 0.02) and inversely associated with fecal glycodeoxycholic acid (r=-0.25, p = 0.03). Participants with NAFLD presented with greater relative abundances of Proteobacteria (p = 0.03), including a trend toward enriched *Sutterella* (p = 0.06), and reduced *Roseburia* (<0.01). Participants with NAFLD presented with higher

abundances of bacteria involved in fat and protein metabolism and depleted abundances of bacteria involved in fiber fermentation, providing evidence for a connection between diet and the gut-microbiota-liver axis. [Hass Avocado Board & USDA National Institute of Food and Agriculture, Hatch Project No. ILLU-698-902.]

2:50

2pBAb6. Quantitative liver fat fraction measurement by multi-view sonography using deep learning and attention maps. Michal Byra (Dept. of Ultrasound, Inst. of Fundamental Technol. Res., Polish Acad. of Sci., Warsaw, Poland), Aiguo Han (Bioacoust. Res. Lab., Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), Andrew S. Boehringer, Yingzhen N. Zhang (Radiology, Univ. of California, San Diego, La Jolla, CA), John W. Erdman (Dept. of Food Sci. and Human Nutrition, Univ. of Illinois at Urbana-Champaign, Urbana, IL), Rohit Loomba (Div. of Gastroenterology, Dept. of Medicine, Univ. of California, San Diego, La Jolla, CA), Mark A. Valasek (Pathol., Univ. of California, San Diego, San Diego, CA), Claude B. Sirlin (Radiology, Univ. of California, San Diego, La Jolla, CA), W. D. O'Brien, Jr. (Bioacoust. Res. Lab., Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Michael P. Andre (Dept. of Radiology, Univ. of California, San Diego, and the San Diego VA Healthcare System, 9500 Gilman Dr., La Jolla, CA 92093, mandre@ucsd.edu)

Qualitative sonography is used to assess nonalcoholic fatty liver disease (NAFLD), an important health issue worldwide. We used B-mode image deep-learning to objectively assess NAFLD in 4 views of the liver (hepatic veins at confluence with inferior vena cava, right portal vein, right posterior portal vein and liver/kidney) in 135 patients with known or suspected NAFLD. Transfer learning with a deep convolutional neural network (CNN) was applied for quantifying fat fraction and diagnosing fatty liver ($\geq$ 5%) using contemporaneous MRI-PDFF as ground truth. Single and multi-view learning approaches were compared. Class activation mapping generated attention maps to highlight regions important for deep learning-based recognition. The most accurate single view was hepatic veins, with area under the receiver operating characteristic curve (AUC) of 0.86 and Spearman's rank correlation coefficient of 0.65. A multi-view ensemble of deep-learning models trained for each view separately improved AUC (0.93) and correlation coefficient (0.76). Attention maps highlighted regions known to be used by radiologists in their qualitative assessment, e.g., hepatic vein-parenchyma interface and liver-kidney interface. Machine learning of four liver views can automatically and objectively assess liver fat. Class activation mapping suggests that the CNN focuses on similar features as radiologists. [No. R01DK106419.]

3:05–3:20 Break

3:20

2pBAb7. Comparison of deep learning and classical breast mass classification methods in ultrasound. Michal Byra (Dept. of Ultrasound, Inst. of Fundamental Technol. Res., Polish Acad. of Sci., Warsaw, Poland), Michael Galperin (Almen Labs., Inc., Vista, CA), Haydee Ojeda-Fournier, Linda Olson, Mary O'Boyle (Radiology, Univ. of California, San Diego, San Diego, CA), Christopher Comstock (Breast Imaging, Memorial Sloan-Kettering Cancer Ctr., New York, NY), and Michael P. Andre (Dept. of Radiology, Univ. of California, San Diego, and the San Diego VA Healthcare System, 9500 Gilman Dr., La Jolla, CA 92093, mandre@ucsd.edu)

We developed breast mass classification methods based on deep convolutional neural networks (CNNs) and morphological features (MF), then compared those to assessment of four experienced radiologists employing BI-RADS protocol. The classification models were developed based on 882 clinical ultrasound B-mode images of masses with confirmed findings and regions of interest indicating mass areas. Various transfer learning techniques, including fine-tuning of a pre-trained CNN, were investigated to develop deep learning models. A matching layer technique was applied to convert gray-scale images to red, green, blue to efficiently utilize discrimination of the pre-trained model. For the classical approach, we calculated MF related to breast mass shape (e.g., height-width ratio, circularity) and then trained binary classifiers. We additionally evaluated both approaches using two publicly available US datasets. Several statistical measures (area under the receiver operating curve [AUC], sensitivity and specificity) were used to assess the classification performance on a test set of 150 cases. The matching layer significantly increased AUC from 0.895 to 0.936 while radiologists' AUCs ranged from 0.806 to 0.882. This study shows both deep learning and classical models achieve high performance. When developed as a clinical tool, the methods examined in this study have potential to aid radiologists accurate breast mass classification with ultrasound.

3:35

2pBAb8. Development and blinded test of a software tool for ultrasound-based hepatic fat fraction estimation. Aiguo Han (Bioacoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 306 North Wright St., Urbana, IL 61801, han51@uiuc.edu), Andrew S. Boehringer, Vivian Montes (Liver Imaging Group, Dept. of Radiology, Univ. of California, San Diego, La Jolla, CA), Michael P. Andre (Dept. of Radiology, Univ. of California, San Diego, La Jolla, CA), John W. Erdman (Dept. of Food Sci. and Human Nutrition, Univ. of Illinois at Urbana-Champaign, Urbana, IL), Rohit Loomba (NAFLD Res. Ctr., Div. of Gastroenterology, Dept. of Medicine, Univ. of California, San Diego, La Jolla, CA), Mark A. Valasek (Dept. of Pathol., Univ. of California, San Diego, San Diego, CA), Claude B. Sirlin (Liver Imaging Group, Dept. of Radiology, Univ. of California, San Diego, La Jolla, CA), and W. D. O'Brien, Jr. (Bioacoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

We have developed quantitative ultrasound (QUS) and deep learning algorithms to estimate hepatic fat fraction from radiofrequency (RF) ultrasound data backscattered by the liver. To facilitate the translation of such algorithms for clinical care and research, we developed a standalone software tool that can automatically generate fat fraction estimates using each of four separate algorithms based on ultrasonic attenuation coefficient (AC) (Algorithm 1), backscatter coefficient (BSC) (Algorithm 2), both AC and BSC (Algorithm 3), and deep learning with uncalibrated raw RF data (Algorithm 4). Reference phantom data and sonographer-drawn fields of interest (FOIs) outlining the liver boundary were used for Algorithms 1–3 but not in Algorithm 4. A pre-determined, fixed FOI was used in Algorithm 4. All four algorithms were developed using contemporaneous MRI-PDFF and ultrasound RF liver data acquired from 144 adult participants with and without nonalcoholic fatty liver disease. The software is now being tested on an independent cohort of participants with contemporaneous MRI-PDFF as the reference standard. 26 participants have been enrolled to date. Preliminary results show good agreement among the four algorithms, and good correlation between the ultrasound fat fraction estimates and MRI-PDFF (Person's $r = 0.61$, 0.63, 0.66, and 0.76 for Algorithms 1–4, respectively). [No. R01DK106419]

3:50

2pBAb9. Comparison of two spectral-based techniques for estimating the attenuation coefficient from human cervix. Ziyang Xu (Bioacoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), Aiguo Han (Bioacoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 306 North Wright St., Urbana, IL 61801, han51@uiuc.edu), Douglas G. Simpson (Dept. of Statistics, Univ. of Illinois at Urbana-Champaign, Champaign, IL), W. D. O'Brien, Jr. (Bioacoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Barbara L. McFarlin (Dept. of Women Children and Family Health Sci., UIC College of Nursing, Univ. of Illinois at Chicago, Chicago, IL)

This study uses *in vivo* radiofrequency ultrasound data acquired from human cervices to compare two commonly used spectral-based techniques for estimating the ultrasonic attenuation coefficient (AC): the spectral difference (SD) and the spectral log difference (SLD) techniques. The AC is a fundamental quantitative ultrasound (QUS) parameter useful for tissue characterization to improve diagnostics, e.g., cervix characterization to predict preterm birth. The selection of appropriate AC techniques is a critical step for QUS tissue characterization. The advantages and disadvantages of various AC estimation techniques have been studied using physical phantoms and computational simulations. However, the heterogeneous nature of real tissue cannot be fully simulated with phantoms and computations. In this study, the SD and SLD techniques were evaluated using human cervices

from 214 pregnant women (enrollment still ongoing). Each participant underwent 1-2 cervical QUS exams during each visit. In each exam, 10 acquisitions of radiofrequency ultrasound and one acquisition of a physical phantom were made using a Siemens MC9-4 transvaginal ultrasound transducer (center frequency: 5.25 MHz) with a Siemens S2000 ultrasound system. Preliminary analysis yielded correlated AC estimates (Pearson's $r = 0.66$; <0.001) between the two AC techniques and better precision (e.g., inter-sonographer reproducibility) with the SLD technique. [No. R01HD089935.]

4:05

2pBAb10. Blood vessel effects in quantitative ultrasound attenuation and backscatter. Andrew S. Boehringer (Liver Imaging Group, Dept. of Radiology, Univ. of California at San Diego, 9452 Medical Ctr. Dr., L3W 501, La Jolla, CA 92037, asboehringer@ucsd.edu), Aiguo Han (BioAcoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), Vivian Montes (Liver Imaging Group, Dept. of Radiology, Univ. of California at San Diego, La Jolla, CA), Tanya Wolfson (Computational and Appl. Statistics Lab., San Diego Supercomputing Ctr., Univ. of California at San Diego, La Jolla, CA), Rohit Loomba (Dept. of Medicine, Univ. of California at San Diego, La Jolla, CA), Michael P. Andre (Dept. of Radiology, Univ. of California at San Diego, La Jolla, CA), John W. Erdman (Dept. of Food Sci. and Human Nutrition, Univ. of Illinois at Urbana-Champaign, Urbana, IL), W. D. O'Brien, Jr. (Bioacoust. Res. Lab., Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL), and Claude B. Sirlin (Liver Imaging Group, Dept. of Radiology, Univ. of California at San Diego, La Jolla, CA)

Nonalcoholic fatty liver disease (NAFLD) is a leading cause of chronic liver disease worldwide. Current diagnosis relies on invasive biopsy. Quantitative ultrasound (QUS) may provide a noninvasive alternative, using parameters such as attenuation (AC) and backscatter (BSC) coefficients. This study examines how exclusion of larger blood vessels from fields of interest (FOI) affects AC and BSC measurements in NAFLD patient livers. 201 total participants with known or suspected NAFLD underwent liver right lobe QUS. Of these, 84 underwent a second QUS exam on the same day and 172 patients had contemporaneous MRI with computation of proton density fat fraction (MRI-PDFF). AC and BSC were calculated from FOIs drawn on reconstructed B-mode images in two ways: FOIs that included as much of the liver as possible including blood vessels and FOIs modified manually to exclude larger blood vessels. Compared to blood vessel-exclusive FOIs, blood vessel-inclusive FOIs provided higher (<0.005) between-exam ICCs (AC, 0.715 vs 0.864; BSC, 0.685 vs 0.796). Spearman's correlation with MRI-PDFF was (AC, 0.422 vs 0.481; BSC, 0.653 vs 0.596). Results suggest that excluding vessels from FOIs makes QUS measurements less repeatable, although relationship with MRI-PDFF did not favor either FOI. [No. R01DK106419.]

4:20

2pBAb11. Frequency content of shear wave pulses excited by acoustic radiation force. Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu)

Ultrasound shear wave elastography uses shear waves to infer the viscoelastic properties of soft tissues. Since soft tissues tend to exhibit dispersive behavior, it is important to standardize the frequency at which measurements are reported. Shear wave pulses excited by acoustic radiation force in different materials, however, do not have the same frequency content, even if the radiation force pulse has precisely the same characteristics. Here, we present a simple model for shear wave pulse creation by acoustic radiation force. This model explains the frequency content of the acoustic radiation force pulse and can therefore be used to design radiation forcing parameters to obtain a desired frequency content in a given material. To derive our model, we use separation of time-scales to show that the shear wave pulse propagation may be treated as an initial value problem. The combination of the (particle) initial velocity distribution and the tissue mechanical properties determines the resulting frequency content of the shear wave pulse.

4:35

2pBAb12. Ultrasound surface wave elastography for assessing extremity lymphedema: A pilot clinical study. Boran Zhou (Radiology, Mayo Clinic, 321 3rd Ave. SW, Rochester, MN 55902, Zhou.Boran@mayo.edu), Samyd Bustos Hemer, Tony Huang (Mayo Clinic, Rochester, MN), Juntao Shao (Radiology, Mayo Clinic, Rochester, MN), Oscar Manrique (Mayo Clinic, Rochester, MN), and Xiaoming Zhang (Radiology, Mayo Clinic, Rochester, MN)

Lymphedema is a condition in which protein-rich fluid accumulates in the tissue causing impaired limb function from swelling and stiffness. Understanding the mechanical properties of soft tissue in lymphedematous limbs is important in clinical diagnosis. We recently developed an ultrasound surface wave elastography (USWE) technique for measuring the shear wave speed of tissue. The aim of this study is to evaluate lymphedematous limb use USWE. A total of 11 patients with lymphedema were included. Both normal and affected extremities were measured. A 0.1-s harmonic vibration was generated by the indenter of the handheld shaker on the limb. Ultrasound probe was used for detecting the tissue motion. Shear wave speed of the tissue was measured using 2-D processing window technique. A region of interest was selected to measure superficial and deep fat tissues. The obtained results showed that the magnitudes of the wave speeds of both the superficial and deep fat tissue at lymphedema sites were statistically higher than those of the control sites, respectively. This suggests that USWE may be useful in assessing correlation between tissue stiffness and lymphedema and would enable physicians to monitor disease severity.

Session 2pEA

Engineering Acoustics: Engineering Acoustics Mixtape: Microfluidics, Arrays, Pipes, Ducts, and Damping

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Roger M. Logan, Cochair
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Contributed Papers

1:15

2pEA1. Axial acoustic field barrier for fluidic microparticle separation at high-throughput. Nan Li (Dept. of Chemical Eng. and Biotechnol., Univ. of Cambridge, Cambridge CB3 0AS, United Kingdom, nl276@cam.ac.uk), Akshay Kale, and Adrian C. Stevenson (Univ. of Cambridge, Cambridge, United Kingdom)

An acoustic field barrier integrated within a flow tubing system to achieve high-throughput separation of particles in fluid will be reported. We investigate the axial acoustic field of a piezo-tube with an inside diameter of 34 mm, a length of 25 mm, and an operating frequency of 1.15 MHz. Energy concentrates within the tube, and leakage at the ends provides a sharp monotonic acoustic pressure field within a fluidic circuit. This process is not the conventional standing wave mechanism; instead, the geometry produces a spatially stable filtering action without fouling. This powerful filtering action is confirmed theoretically via a COMSOL simulation and demonstrated experimentally by concentrating suspensions of 5 μm proteoglycan tracer particles at a flow rate of 20 ml/min: The corresponding acoustic contrast factor is 0.243, and the trapping force is 11 pN. This tube geometry tackles the limitations of microfluidic standing wave based acoustic concentrators, namely, complex extraction, low-throughput, and distributed focus, by harnessing a stable monotonic field profile.

1:30

2pEA2. MHz-order surface acoustic wave thruster for underwater silent propulsion. Naiqing Zhang (Mech. and Aerosp. Eng., UCSD, Structural and Mater. Eng. (SME) Bldg. Rm. 320, Matthews Ln, La Jolla, CA 92093, naz016@eng.ucsd.edu), Yue Wen, and James Friend (Mech. and Aerosp. Eng., UCSD, La Jolla, CA)

High frequency surface acoustic waves with MHz order are able to generate intense acoustic streaming flow transformed from the attenuation of acoustic radiation pressure in viscous fluid. We here demonstrate the fluid propulsion force onto the device of SAW chip as a reaction force based on the change of momentum flux of acoustic streaming flow. A pendulum force balancing method is established to simply quantify 40 MHz SAW propulsion force onto lithium niobate (LN) chip. Propulsion force with 5 mN can be easily generated by applying ~1 W power to SAW device. A theoretical model, along with acoustic streaming profile via particle image velocimetry, is proposed to explain the propulsion force based on applied power and different fluid viscosity. Finally, we propose a model optimizing LN chip size to perform maximum propulsion force per unit device area. The maximum force of ~ 10 mN/cm^2 with 1 W power using 40 MHz SAW has been performed in water. With high-frequency, short attenuation length, relatively low applied power and large propulsion force per surface area, it shows great potential of SAW-induced thruster for underwater silent propulsion and transportation.

1:45

2pEA3. Nanoacoustofluidics: Unprecedented fluid manipulation via controlled MHz-order vibration at the nanoscale. Naiqing Zhang (Mech. and Aerosp. Eng., Univ. of California San Diego, Structural and Mater. Eng. (SME) Bldg. Rm. 320, Matthews Ln, La Jolla, CA 92093, naz016@eng.ucsd.edu), Ofer Manor (Wolfson Dept. of Chemical Eng., Technion-Israel Inst. of Technol., Haifa, Israel), and James Friend (Mech. and Aerosp. Eng., Univ. of California San Diego , La Jolla, CA)

Manipulation of fluids and colloids at the nanoscale is made exceptionally difficult by the dominance of surface and viscous forces. The use of MHz-order vibration has dramatically expanded in microfluidics, enabling fluid manipulation, atomization, and microscale particle/cell separation. We find even more powerful results at the nanoscale, with the key discovery of a new mechanism of acoustic wave-fluid motion interaction. We show that MHz-order surface acoustic waves (SAW) can manipulate fluids and fluid droplets within fully transparent, high-aspect ratio 5–150 nm tall, 10–100 μm wide, 5 mm long nanoslits fabricated via a direct, room temperature bonding method for lithium niobate (LN). The application of SAW causes fluid draining and pumping, developing ~1 MPa pressure through the nanoslit structure. The mechanism of specific draining pattern is well explained by a detailed theoretical model based on the nonlinear acoustic vibration phenomenon in nanoscale channels. We also show how individual fluid droplets of only 1–10 fl are propelled by incident SAW in the nanoslit structure, entrapping them at locally widened regions along the nanoslit due to the capillary pressure difference. Our results indicate MHz-order SAW as a powerful tool for fluid manipulation in nanoscale.

2:00

2pEA4. Immersive wave control experimentation using compensated directive sources. Xun Li (Inst. of Geophys., ETH Zurich, NO H 41.2, Sonneggstrasse 5, Zurich CH-8045, Switzerland, lixunjack@outlook.com), Nele Börsing, Theodor S. Becker, Dirk-Jan van Manen (Inst. of Geophys., ETH Zurich, Zürich, Switzerland), Andrew Curtis (School of GeoSci., The Univ. of Edinburgh, Edinburgh, United Kingdom), and Johan O. Robertsson (Inst. of Geophys., ETH Zurich, Zürich, Switzerland)

A physical boundary with embedded active sources can cancel acoustic waves incident on the boundary and also synthesize waves to fully control the acoustic field in an experimental setup (e.g., a water tank) such that the physical experiment is artificially immersed into a virtual domain and waves propagate seamlessly between the physical experiment and virtual domain. The acoustic representation theorem at the heart of the immersive experiment requires physical monopole sources to be deployed on the active boundary. However, real physical sources (e.g., piezoelectric transducers) project waves at sonic frequencies (e.g., 2–20 kHz) that do not fully conform to the theoretically required radiation pattern. If left uncorrected, using

these physical sources causes the wavefield to deviate from those desired in immersive experiments. A method is proposed to compensate for the non-monopolar radiation pattern of the sources, and the compensation is incorporated into the Kirchhoff-Helmholtz extrapolation, which is used to determine the controlling field at the active boundary in real-time. Numerical simulations show that the method can effectively compensate for the undesired effects caused by such sources. The method is implemented as a preprocessing step that modifies the extrapolation Green's functions in the Kirchhoff-Helmholtz integral before the actual experiments take place and can be physically interpreted in terms of Huygens' principle.

2:15

2pEA5. Reproduction of sound field dynamics by considering virtual acoustic impedance. Eiji Yokota (Integrated design Eng., Keio Univ., 3-14-1, Hiyoshi, Kohoku, Yokohama, Kanagawa 223-8522, Japan, yokota@katsura.sd.keio.ac.jp) and Seiichiro Katsura (Integrated Design Eng., Keio Univ., Yokohama, Japan)

Research on sound field reproduction mainly focuses on sound signal. However, the dynamics of a sound field needs to be taken into account when sound environment information (e.g., position of walls) in the original sound field changes. This paper proposes an acoustic control system that reproduces the dynamics of the original sound field by controlling the sound source that is a loudspeaker. Sound environment information is represented as virtual acoustic impedance, which is calculated with sound propagation models that regard a reflection surface as a mass-spring- damper system. The output of the control sound source is determined by recreating the virtual acoustic impedance of a specific position in the original sound field. This enables sound environment information of the original sound field to be reproduced in reconstructed sound fields. Therefore, the proposed method is able to deal with the situation where the dynamics of sound fields changes. Finally, to validate the proposed method, a one-dimension sound field with a rubber wall as the sound environment is reproduced using the proposed method.

2:30

2pEA6. A uniform rectangular array of isotropic sensors of stochastic gains: The hybrid Cramer-Rao bound for direction finding. Dominic M. Kitavi (Dept. of Mathematics, Computing, and Information Technol., Univ. of Embu, Embu, Kenya) and Kainam T. Wong (School of General Eng., Beihang Univ., 37 Xueyuan Rd., Haidian District, Beijing 100083, China, ktwong@ieee.org)

Real-world sensors deviate from their nominal gain; that deviation could be modeled stochastically. Such sensors, populating a rectangular array grid, would affect the consequential direction finding's statistical precision, which is quantified here via the Cramer-Rao lower bound of the direction-of-arrival estimation variance.

2:45–3:00 Break

3:00

2pEA7. Time-reversal techniques for defect detection in pipe system using high frequency acoustic waves. Saber Nasraoui (Dept. of Civil and Environ. Eng., Hong Kong Univ. of Sci. and Technol., Res. Staff Quarter HKUST, Sau Kung New Territories, Clear Water Bay Rd. 660, Hong Kong, nasraouisaber@yahoo.fr), Moez Louati, Mehmet Murat Gozum, George Grigoropoulos, and Mohamed S Ghidaoui (Dept. of Civil and Environ. Eng., Hong Kong Univ. of Sci. and Technol., Sai Kung, New Territories, Hong Kong)

Recent research has shown that transient (acoustic)-based defect detection methods (TBDDM) in water pipelines are a promising new approach that may help achieve more sustainable urban water supply systems (UWSS). These methods are based on the generation of a probing pressure wave in the pipe system, and the use of the measured response to identify defects. This paper develops an analytical solution for sound propagation in inviscid axisymmetric pipe system with sound source at the boundary. The solution is given for rigid and elastic pipe wall. This analytical model is used to apply time reversal and match-field processing to the detection of blockages and leaks in a single pipe system. The wave forms used at the source are Gaussian-modulated sine wave and Chirp with frequency ranging from 5 to 60 kHz. The results show accurate detection with resolution of the order of the probing wave length.

3:15

2pEA8. Experimental validation of time-reversal technique for defect detection in air-filled pipes using high frequency acoustic waves. Mehmet M. Gozum (Civil and Environ. Eng., Hong Kong Univ. of Sci. and Technol., HKUST, Kowloon 100025, Hong Kong, muratgozum19@gmail.com), Moez Louati (Civil and Environ. Eng., Hong Kong Univ. of Sci. and Technol., Sai Kung, New Territories, Hong Kong), Saber Nasraoui (Civil and Environ. Eng., Hong Kong Univ. of Sci. and Technol., Sau Kung New Territories, Clear Water Bay Rd., Hong Kong), George Grigoropoulos (Civil and Environ. Eng., Hong Kong Univ. of Sci. and Technol., Sai Kung, New Territories, Hong Kong), and Mohamed S Ghidaoui (Civil and Environ. Eng., Hong Kong Univ. of Sci. and Technol., Sai Kung, New Territories, Hong Kong)

Pipeline systems are extensively used for the transportation and distribution of the gas, water and petroleum products. The sustainability of the pipelines can be improved by utilizing the transient (acoustic)-based defect detection methods (TBDDM). In these methods, the probing pressure wave is generated in the pipe system, and the measured response is used to identify the defects. In this paper, experiments are performed on the acrylic pipe to study the acoustic wave propagation. The acrylic pipe is filled with air such that rigid pipe assumption is valid. Theoretical predictions applied with time reversal and match-field processing are validated with the experiments for the leak detection in a pipe system. The waveforms of Gaussian-modulated sine wave and Chirp are utilized as the transmitted signals in the frequency range of 1–15 kHz. The experimental results show that time-reversal technique is capable of accurate detection with a good resolution.

3:30

2pEA9. Theoretical considerations on low-frequency wave propagation in a duct of variable cross section with impedance exit conditions. Juhyeong Cho (Korea Inst. of Machinery and Mater., Jang Dong 171, Daejeon 34103, South Korea, antocho@kimm.re.kr)

Acoustic characteristics in ducts of variable cross section have been gaining extensive attention as they have been used in various applications such as mufflers, tunable acoustic resonators, etc. The present study deals with theoretical analysis of low-frequency wave propagation in a duct of variable cross sectional area with arbitrary impedance exit conditions to formulate the complex amplitudes of the reflected and/or transmitted waves and examine how the reflection/transmission coefficient at the area expansion/constriction is affected by the exit boundary condition and the dimensions such as the cross sectional area ratio and the axial length. Some of the results, for instance, show that, whenever the magnitude of the exit reflection coefficient is one ($|R_{exit}| = 1$), e.g., for a closed or an open exit, the magnitude of the reflection coefficient at the area change is also conserved (= 1) regardless of the dimensions of a duct while the phase of the reflection coefficient varies depending on its dimensions. Attainable ranges of the reflection coefficient at the area change are also evaluated depending on its dimensions and the exit impedance. This study will provide useful formulations and insights in the design of a noise attenuator or a resonator, for instance, with an adjustable-length partial blockage in it.

3:45

2pEA10. Optimal damping in ducts—The Cremer impedance. Mats Åbom (The Marcus Wallenberg Lab., KTH-The Royal Inst of Technol., Teknikringen 8, Stockholm 10044, Sweden, matsabom@kth.se) and Zhe Zhang (The Marcus Wallenberg Lab., KTH-The Royal Inst of Technol., Stockholm, Sweden)

The Cremer impedance, first proposed by Cremer (1953) and then extended by Tester (1973), is supposed to give the maximum propagation damping in an infinitely long waveguide. Previous works including a uniform grazing flow have shown negative resistance in the low frequency range for both circular and 2-D rectangular waveguides, i.e., implying an active boundary. In order to further analyze the low frequency behavior of the Cremer impedance, especially the negative resistance, two investigations

are conducted in the current work. First, the previously used Ingard-Myers boundary condition is replaced by the Brambley boundary condition with the introduction of a thin inviscid boundary layer, and results obtained with the two boundary conditions are compared to see the effect of a sheared flow. Second, discussions regarding the validity of the low frequency result in both the up- and downstream directions from the perspective of mode merging are presented. This analysis is further extended from the fundamental mode to higher order modes in the frequency range where they are "just cut-on."

4:00

2pEA11. Experimental assessment of the sound absorption coefficient of three natural fibers. Ana Carolina M. Mansur, Alisson Zanetti, Nilson Barbieri (Pontifícia Universidade Católica do Paraná, Curitiba, Paraná, Brazil), and Key F. Lima (Pontifícia Universidade Católica do Paraná, Imaculada Conceição, 1155, Curitiba, Paraná 80215901, Brazil, keyflima@gmail.com)

The acoustic absorptive materials used in noise control stand out for the high efficiency in sound absorption. However, many of these materials are synthetic and hence they have a long or indeterminate decomposition time and does not allow their disposal in nature. In addition, if these materials suffer other destination as incineration in industrial furnaces, they may release toxic gases harmful to health and the environment. Natural fibers are an alternative for the replacement of synthetic materials used in acoustic insulation or in the reverberating surfaces treatment. The main objective of this work is to assess the acoustic efficiency of sisal, coconut husk and sugarcane bagasse fibers in terms of their sound absorption coefficient. The samples were experimentally assessed with the 100 mm diameter impedance tube according to the ASTM C1050-12 standard. The samples evaluated have 30, 40, and 50 mm thick and were tested with and without air gap between the samples and the back rigid of the tube in low and medium frequencies. Moreover, perforated plates were introduced in front of the samples to increase their efficiencies.

4:15

2pEA12. Optimal damping for a bend containing a microperforated panel. Cheng Yang (School of Mech. Eng., Shanghai Jiao Tong Univ., Shanghai 200240, China, cheng.yang@sjtu.edu.cn), Stefan Sack (The Marcus Wallenberg Lab. for Sound and Vib. Res., KTH Royal Inst. of Technol., Stockholm, Sweden), and Mats Åbom (The Marcus Wallenberg Lab. for Sound and Vib. Res., KTH Royal Inst. of Technol., Stockholm, Sweden)

Microperforated panels (MPPs) with a backing cavity can be used as liners on duct walls to damp acoustic waves. However, recent studies show that a MPP (without the backing cavity) placed inside a duct also could give rise to an effective damping. But only if there is an acoustic pressure gradient across the MPP, e.g., as in a duct bend. This offers a new way of designing MPP for control of duct acoustics and leads to a variety of potential applications. In particular, the widely used guide vanes in a bend might be replaced by MPPs which serve as an acoustic damping treatment, while at the same time maintain the aerodynamic performance. In this work, the optimal damping by a MPP in a bend will be investigated based on the so called Cremer impedance theory. A derivation of the governing equation associated with this optimal value will be made for this configuration and the procedure for solving the equation presented. The trajectory of the axial wavenumber in the complex domain is investigated as well as the resulting optimal damping and the Cremer impedance. The results will be compared with the corresponding Cremer solution for a straight duct. Finally, possibilities to realize the Cremer impedance required for the optimal damping in a bend is discussed.

4:30

2pEA13. Demonstration study of applying bias-flow perforated liner on damping self-excited thermoacoustic oscillations in T-shaped thermoacoustic combustor. Dan Zhao (Dept. of Mech. Eng., Univ. of Canterbury, Private Bag 4800, Christchurch 8140, New Zealand, dan.zhao@canterbury.ac.nz), Di Guan (Dept. of Mech. Eng., Univ. of Canterbury, Christchurch, New Zealand), and A. Tarique (Nanyang Technol. Univ., Singapore, Singapore)

Self-excited thermoacoustic instability is undesirable in propulsion and power generation systems due to its detrimental effects. In this work, we conducted experimental demonstration study of applying bias-flow perforated liners on attenuating self-excited thermoacoustic oscillations in a premixed T-shaped combustor. For this, two perforated liners with different porosities and variable bias flow rate are experimentally tested and compared. It is found that in the absence of bias flow (cooling flow), the premixed flame can produce and sustain limit cycle oscillations. The amplitude and frequency of the dominant thermoacoustic mode are approximately 215 Hz and 140 dB. However, as the cooling flow velocity is increased to 5.5 m/s, the limit cycle oscillations are completely attenuated. More than 50 dB sound pressure level reduction is achieved. High-frequency modes are dampen by more than 25 dB. Further experimental test is conducted on the perforated liner with a porosity of 2.9%. Similar attenuation effect is achieved. Finally, during the transient growth process, dominant mode switching from high (non-harmonic) to low frequency is observed. The present work shed lights on the optimum design of perforated liner on attenuating thermoacoustic instability and the dynamic behaviours of the transient growth process.

Session 2pIDa

Interdisciplinary and Student Council: Guidance From the Experts: Applying for Grants and Fellowships

Daniel Guest, Cochair
Department of Psychology, University of Minnesota, 75 E River Road, Minneapolis, Minnesota 55455

Eric Rokni, Cochair
Penn State University, 201 Applied Science Building, The Pennsylvania State University, State College, Pennsylvania 16801

A panel of successful fellowship winners, selection committee members, and fellowship agency members will answer questions regarding grants and fellowships, application advice, and funding opportunities. The panelists will briefly introduce themselves, followed by a question and answer session with the audience.

TUESDAY AFTERNOON, 3 DECEMBER 2019 CAROUSEL, 2:40 P.M. TO 5:10 P.M.

Session 2pIDb

Interdisciplinary: Introduction to Technical Committees

Kieren H. Smith, Cochair
Arizona State Univ., Tempe, Arizona 85287

William Doebler, Cochair
NASA Langley Research Center, Mail Stop 462, Hampton, Virginia 23681

Alexandra M. Padilla, Cochair
School of Marine Science and Ocean Engineering, University of New Hampshire, Forest Park Apt. 281, Durham, New Hampshire 03857

Chair's Introduction—2:40

Invited Papers

2:45

2pIDb1. Introduction to the Technical Committee on Engineering Acoustics. Michael R. Haberman (Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu)

This talk will provide an introduction to ongoing work in the Technical Committee on Engineering Acoustics (TCEA) of the Acoustical Society of America, which is one of the most diverse Technical Committees of the Society. Engineering Acoustics encompasses the theory and practice of creating tools for investigating acoustical phenomena and applying knowledge of acoustics to practical utility. This includes the design and modeling of acoustical and vibrational transducers, arrays, and transduction systems in all media and frequency ranges. It is also concerned with the design of acoustical instrumentation, metrology, and the calibration of those systems. It further considers all aspects of measurement and computational techniques as they relate to acoustical phenomena and their utility. The talk will provide an introduction to a broad range of research topics in TCEA, with specific emphasis on exciting new areas of research.

2:55

2pIDb2. Introduction to the Technical Committee for Noise. James E. Phillips (Wilson Ihrig, 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

Noise presents both a health hazard and a societal hazard. Community noise, product noise, noise generated within the built environment, and noise within the work environment have affected where and how we interact with the soundscapes where we live and work. The United States Environmental Protection Agency, Office of Noise Abatement and Control was responsible for noise regulations that rate the production and reduction of noise until it was defunded. Federal agencies such as the National Institute for Occupational Safety and Health, the Occupational Safety and Health Administration and the Mine Safety and Health Administration have conducted research to develop regulations for occupational noise exposures. Recent community noise issues have focused on windfarms and the low-frequency noise that nearby residents might experience. This presentation will cover a wide range of noise-related topics and highlight efforts in standards for noise assessment and noise control.

3:05

2pIDb3. Structural Acoustics and Vibration: Which came first, the sound or the vibration? Robert M. Koch (Chief Technol. Office, Naval Undersea Warfare Ctr., Code 1176 Howell St., Bldg. 1346/4, Code 01CTO, Newport, RI 02841-1708, Robert.M.Koch@navy.mil)

Structural acoustics may be defined as a multidisciplinary coupled field of physics usually referring to the characterization of either (1) the sound power emitted by a vibrating structure subjected to external dynamic excitation or (2) the vibrational response of structures excited by incident sound fields or fluid excitation. The Structural Acoustics and Vibration (SAV) Technical Committee is a diverse group of scientists and engineers sharing a common interest and shared stewardship in the health and advancement of the SAV technical discipline. In addition to the fundamental scientific study of the underlying SAV physics which is important to all SAV practitioners, there is also interest on the practical real-world, application of the SAV science to predict, control, and potentially reduce the vibroacoustic response of given structural acoustic systems. This paper begins by providing a fundamental definition of the underlying physics governing the SAV technical discipline. The numerous categories and subdivisions within the general SAV area are then addressed, along with illustrations of the broad array of scientific and engineering real-world applications in the field. Following that, examples of interesting career options as well as exciting new research areas in SAV are presented. Finally, this paper concludes with information regarding the makeup, functioning, and administrative philosophy of the SAVTC.

3:15

2pIDb4. Biomedical Acoustics: Breaking old barriers in science and medicine. Subha Maruvada (U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993, subha.maruvada@fda.hhs.gov)

The Technical Committee on Biomedical Acoustics (BATC) is one of the most diverse groups in the Acoustical Society of America. BATC is comprised of scientists and engineers who study a wide range of biomedical applications using ultrasound from imaging fetuses to ablating fibroids and treating brain disorders. Diagnostic ultrasound has expanded to include exciting new techniques such as shear wave elastography and opto-acoustic imaging. Therapeutic ultrasound applications include physiotherapy, lithotripsy, as well as the treatment of tumors via thermal ablation, i.e., HITU, or mechanical ablation, i.e., histotripsy. The most recent developments in therapeutic ultrasound involve neuromodulation and transcranial magnetic resonance guided focused ultrasound for the treatment of various brain disorders, including essential tremor, neuropathic pain, and Parkinson's disease. The ability to temporarily open the blood-brain barrier using focused ultrasound has been demonstrated in animals and, more recently, in humans. Opening of the blood-brain barrier could allow the transmission of drugs to treat brain diseases. Diagnostic and therapeutic ultrasound applications continue to advance the field of medicine and are integral to the advancement of both diagnosis and treatment of debilitating diseases.

3:25

2pIDb5. Signal Processing: Data analysis, machine learning, and imaging sources. Brian E. Anderson (Dept. of Phys. & Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, bea@byu.edu)

Whether it is locating an enemy submarine, finding that annoying sound generated inside an airplane, optimizing the focusing of sound to destroy kidney stones, or learning how bats track their food, advanced signal processing is key to the success of these research topics. The Technical Committee on Signal Processing in Acoustics was formed to foster interdisciplinary interaction among the Technical Committees. Techniques used to track a submarine in the ocean, should also work to track a swarm of bees flying through the air or to track the delivery of medicine to a targeted place in the body. For example, at the previous ASA meeting a signal processing session on machine learning brought together researchers using machine learning to identify earthquakes from the global smartphone seismic network, to determine whether noise is further endangering a certain species of birds, and to map out the ocean floor. This presentation will highlight some of the exciting techniques being used in signal processing and in the process give an appreciation for the breadth of the applications of the techniques being developed.

3:35–3:50 Break

3:50

2pIDb6. An Introduction to the Technical Committee on Animal Bioacoustics. Marla M. Holt (NOAA NMFS NWFSC, 2725 Montlake Blvd East, Seattle, WA 98112, Marla.Holt@noaa.gov)

Animal Bioacoustics as a field of research involves the study of sound in non-human animals. The range of this field is wide and includes all aspects of sound production and reception, communication and associated behaviors, acoustic ecology and effects of noise/sound, and passive and active acoustic methods for monitoring individuals, populations, habitats and ecosystems. Members of the Technical Committee on Animal Bioacoustics (TCAB) come from diverse backgrounds, including those with training in biology, ecology,

engineering, mathematics, oceanography, physics, and psychology. Many TCAB members also participate in the activities of other ASA Technical Committees including Acoustical Oceanography, Engineering Acoustics, Noise, Psychological and Physiological Acoustics, Signal Processing, and Underwater Acoustics, reflecting the interdisciplinary nature of the field. As Chair, I will highlight the history of TCAB, some of the popular and emerging areas of research in Animal Bioacoustics, and the ways that ASA members can get involved in this exciting area of acoustics.

4:00

2pIDb7. Architectural Acoustics: From Auditoria to Zoos. Ana M. Jaramillo (Olson Sound Design, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu)

The Technical Committee in Architectural Acoustics deals with sound in the built environment. Any type of building requires consideration of sound during the design process. In some cases, sound quality is very important (e.g., concert halls), in others, sound clarity is essential for speech communication (e.g., classrooms) and in some, the presence of unwanted sound can be detrimental (e.g., hospitals). All of these scenarios are considered by architectural acoustics. Because of the broad range of applications, TCAA often crosses interests with other Technical Committees. Noise, Musical Acoustics, Psychological and Physiological Acoustics are some of the most common ones to hold joint sessions with and have research and application interests in common. As TCAA is one of the largest ones in the Society, it also has a great number of subcommittees dedicated to specific topics such as Classroom Acoustics or Green Buildings. The committee members are also varied, with a good mix of academics, industry and practitioners, as well as students.

4:10

2pIDb8. An introduction to the Acoustical Oceanography Technical Committee. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu) and Grant B. Deane (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

Acoustics is a critical modality for interpreting the ocean environment, as evidenced by the complex acoustic physiology and sophisticated auditory processing capability of marine organisms, for example. The acoustical oceanography (AO) technical committee (TC) seeks to foster a broad range of work in pure and applied acoustics aimed at gaining new and fundamental understanding of physical, biological, geophysical, and chemical processes in the ocean and at its boundaries. With this focus, the AO TC is necessarily strongly interdisciplinary, having close ties to other TC's such as animal bioacoustics, physical acoustics, signal processing, and underwater acoustics. This talk will describe many of the break-through discoveries that have been made in our field and point to inspiring present and future work.

4:20

2pIDb9. Overview of Speech Communication Research. Linda Polka (School of Commun. Sci. & Disord., McGill Univ., 2001 McGill College Ave., 8th Fl., SCSD, McGill University, Montreal, PQ H3A 1 G1, Canada, linda.polka@mcgill.ca)

Although speech communication is a fundamental part of our daily behavior, it is not yet well understood. Speech communication research examines how spoken language is produced, transmitted and perceived. This involves a number of different disciplines, from linguistics and experimental psychology to speech and hearing sciences, and electrical engineering. The field covers a wide range of physiological, psychological, acoustic, and linguistic phenomena. In this talk, I will provide a broad overview of the types of research that happen within our TC. I will highlight the advantages of participating in the ASA through examples that show how work in Speech Communication may inform or be informed by work in other technical areas within our society.

4:30

2pIDb10. The Technical Committee on Musical Acoustics: Integral part of the interdisciplinary nature of acoustics. Andrew C. Morrison (Joliet Junior College, 1215 Houbolt Rd., Natural Sci. Dept., Joliet, IL 60431, amorrison@jjc.edu)

The Technical Committee on Musical Acoustics (TCMU) of the Acoustical Society of America (ASA) is concerned with the application of science and technology to the field of music. Traditionally, the areas of interest for those doing musical acoustics have included: the physics of musical sound production, questions of music perception and cognition, and the analysis and synthesis of musical sounds and composition. It is clear that many of the technical committees of the ASA are highly interdisciplinary and TCMU is no exception. The TCMU organizes special sessions often featuring a family of musical instruments, a style of music, or a technique of musical performance. Additionally, sessions have featured innovative techniques for studying acoustics which have broad application across the other technical committees. TCMU also regularly hosts musical performances at ASA meetings and visits to local music venues or musical instrument factories in order to connect researchers to practitioners. The TCMU encourages all members of the ASA to engage with us in the pursuit of understanding the production, radiation, perception, and cognition of musical sounds.

4:40

2pIDb11. Psychological and Physiological Acoustics: From sound to sensation. Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

The area of psychological and physiological acoustics encompasses a wide and multidisciplinary range of topics. It is concerned with questions of what happens to sound once it enters the auditory system, and how sound is processed to facilitate communication and navigation. Topics include the biomechanics of the middle and inner ear; the neuroscience of the auditory nerve, brainstem, and cortex; and behavioral studies of auditory perception and cognition. This presentation will provide an overview of some of the many areas currently under investigation, ranging from basic questions about the neural representations of different sound features to clinical applications, such as the development and improvement of hearing aids, as well as cochlear, brainstem, and even midbrain implants that bypass the peripheral auditory system to provide some hearing to people with profound hearing loss. [Work supported by NIH Grant No. R01 DC012262.]

4:50

2pIDb12. An introduction to research topics in Underwater Acoustics. Jason D. Sagers (Environ. Sci. Lab., Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sagers@arlut.utexas.edu)

The Acoustical Society of America Technical Committee on Underwater Acoustics (UW) investigates sound wave phenomena in marine environments, including oceans, lakes, and rivers. Research interests span a broad spectrum from the measurement and modeling of acoustic propagation and scattering, to the detection and characterization of underwater sound, to signal processing algorithms and statistics. This diverse technical committee also shares interests with Animal Bioacoustics (AB), Acoustical Oceanography (AO), and Signal Processing (SP). This talk will highlight a few past and present research topics in underwater acoustics, emphasizing the important role of sound as a tool in subsea research and exploration.

5:00

2pIDb13. The Technical Committee on Physical Acoustics: Who are we and what do we do? Kent L. Gee (Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu) and Veerle M. Keppens (Univ. of Tennessee, Knoxville, TN)

The Technical Committee on Physical Acoustics (TCPA) of the Acoustical Society of America (ASA) includes scientists and engineers interested in the underlying physics of acoustical phenomena or in using acoustic waves to study the physical properties of matter. TCPA is broad: computational, experimental, and analytical methods are used across the entire frequency range—from infrasound through ultrasound—to study wave propagation through liquids, solids, and gases. This presentation introduces new students to some of the facets of TCPA and its intersection with other ASA technical areas.

TUESDAY AFTERNOON, 3 DECEMBER 2019 CONTINENTAL/CRYSTAL, 1:00 P.M. TO 2:25 P.M.

Session 2pNSa

Noise, Architectural Acoustics, and ASA Committee on Standards: Current Trends and Advancements in Applying Acoustics to Smart Cities

Brigitte Schulte-Fortkamp, Cochair
Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 10587, Germany

Bennett M. Brooks, Cochair
Brooks Acoustics Corporation, 49 N. Federal Highway, Unit 121, Pompano Beach, Florida 33062

Chair's Introduction—1:00

Invited Papers

1:05

2pNSa1. Soundscape and smart growth. Bennett M. Brooks (Brooks acoust. Corp., 49 N. Federal Hwy., Unit 121, Pompano Beach, FL 33062, bbrooks@brooksacoustics.com) and Brigitte Schulte-Fortkamp (Tech. Univ. of Berlin, Berlin, Germany)

Smart growth principles are increasingly being applied to urban developments, in order to increase the quality of life for residents, workers and visitors in a locale. A key component which contributes to the personal experiences of those occupants is the acoustical environment. The soundscape technique is a powerful tool for determining existing conditions, and for developing positive design and planning outcomes. Therefore, it is important that soundscape be incorporated with smart growth principles, to achieve the full potential for improvement in each development. A review of smart growth principles is presented. The relationship of these principles to soundscape analysis is examined.

1:25

2pNSa2. Making cities smarter with new soundscape indices. Andrew Mitchell (Inst. for Environ. Design and Eng., Univ. College London, Central House, 14 Upper Woburn, London WC1H 0NN, United Kingdom, andrew.mitchell.18@ucl.ac.uk), Francesco Aletta, Tin Oberman, Mercede Erfanian, Magdalena Kachlicka, Matteo Lionello, and Jian Kang (Inst. for Environ. Design and Eng., Univ. College London, London, United Kingdom)

The core objectives of smart city design are to increase quality of life, enhance efficiency, and move towards the sustainability of cities. While this will involve increased integration of new and smarter technologies into urban design, the implementation of these technologies as applied to acoustics should be made within a design approach which considers these core objectives. Soundscape strategies have a focus on people's perception and experience, considering the many factors which influence their perception. A recognized demand in the field of soundscape is a new set of metrics that can reliably measure both the acoustic environment and its perception. The European Research Council acknowledged this need and recently funded the Advanced Grant project "Soundscape Indices" (SSID), which aims at providing more advanced tools, compared to conventional dB-based metrics, by taking into account psychological, (psycho)acoustical, neurophysiological, and contextual factors for soundscape assessment. The SSID project will: characterise soundscapes, by capturing acoustic environments and establishing a comprehensive database; identify key factors and their influence on soundscape quality based on the database, by conducting laboratory psychological evaluations, acoustical/psychoacoustic factors analysis; and research the neural and psychophysiological underpinnings of soundscape experience. It is expected that SSID will provide a vital tool in guiding the implementation of the technological infrastructure of smart cities.

1:45

2pNSa3. Untapping the potential of soundwalks as participatory methods for co-designing smart cities. Antonella Radicchi (Institut für Stadt- und Regionalplanung, Technische Universität Berlin, Hardenbergstraße 40 a, Sekr. B 4, Berlin 10623, Germany, antonella.radicchi@tu-berlin.de)

Nowadays, the mainstream smart city paradigm relies on uncritical, massive use of technology, deployed with the promise of addressing sustainability challenges affecting large, densely populated cities. Likewise, a scrutiny of literature in the field of environmental noise shows the tendency to exploit technological innovation to implement a noise-based, top-down approach in the evaluation of the acoustic environment, which on the other hand overlooks the health and psychological effects of noise on people. This contribution reflects on this contradiction and discusses how the people-centred soundscape concept and its methods, i.e., the soundwalks, can counterbalance such criticalities informing the smart city paradigm. After providing a brief introduction to soundscape theory and methods, two case studies of soundwalks conducted with the public in Berlin and New York are presented. In conclusion, the studies' limitations are discussed and recommendations on the potential of soundwalks as participatory methods for co-designing smart cities are provided so as to possibly orientate future research and professional practice.

2:05

2pNSa4. Comparing soundscape assessment methods with natural-language expressions and a questionnaire survey: A case study in Seoul, Korea. Jin Yong Jeon (Dept. of Architectural Eng., Hanyang Univ., Seoul 04763, South Korea, jyjeon@hanyang.ac.kr) and Hyun In Jo (Architectural Eng., Hanyang Univ., Seoul, Seongdong-gu, South Korea)

A Swedish soundscape quality protocol was developed and has been used as a representative soundscape-quality assessment method around Europe. However, this protocol has limitations in Asian countries because the countries have distinctly different social and cultural backgrounds and form their own unique soundscapes. Consequently, this study examined the utility of a soundscape assessment survey for Koreans by comparing unconstrained natural-language expressions and existing protocol survey questions. To this end, soundscape data from a variety of places, e.g., parks, tourist attractions, and commercial areas in Seoul, Korea, were collected using 360-deg cameras and Soundfield microphones. In addition, a virtual-reality (VR) assessment environment was constructed using a head-mounted device (HMD) and 3-D audio technology under laboratory conditions, and the collected soundscapes were assessed. The participants were encouraged to freely and verbally express their thoughts about the sound environment of each place, and text data were obtained through voice-recognition technology. The protocol survey was conducted for the same subjects after the assessment. The results showed that it is necessary to develop a soundscape assessment standard that is appropriate for Koreans because differences from the existing European-style soundscape protocol were revealed in the assessment items, when an emotional analysis was conducted using text-mining technology.

Session 2pNSb

Noise: General Noise Session

William J. Murphy, Chair
Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, Ohio 47025

Contributed Papers

3:00

2pNSb1. 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 and 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 and personnel and community noise abatement measures are identified for these indoor and outdoor firing ranges.

3:15

2pNSb2. The temporary hearing thresholds shifts in fitness instructors. Adam Dudarewicz (Dept. of Physical Hazards, Nofer Inst. of Occupational Medicine, Lodz, Poland), Kamil Zaborowski (Dept. of Physical Hazards, Nofer Inst. of Occupational Medicine, 8 Sw. Teresy Str., Lodz 91-348, Poland, kamil.zaborowski@imp.lodz.pl), Malgorzata Pawlaczyk-Luszczynska (Dept. of Physical Hazards, Nofer Inst. of Occupational Medicine, Lodz, Poland), Anna Wolniakowska (Clinic of Audiol. and Phoniatrics, Nofer Inst. of Occupational Medicine, Lodz, Poland), and Mariola Sliwinska-Kowalska (Clinic of Audiol. and Phoniatrics, Nofer Inst. of Occupational Medicine, Lodz, Poland)

Noise in the entertainment industry often reaches high sound pressure levels. The aim of this study was to analyze the temporary threshold shifts (TTSs) of hearing among fitness instructors. The study comprised a total of 30 fitness instructors working in fitness clubs. The noise dosimeters were used to determine individual noise exposure during exercises conducted by the instructors. To assess the TTS values, the pure tone audiometry tests were performed before and just after the exercises. The observed TTSs were compared with the predicted TTSs according to the computational model developed by Melnick (1991). Typical fitness exercises lasted from 1 to 2 h and A-weighted equivalent-continuous sound pressure level ranged from 75 to 96 dB. Temporary threshold shifts after the sound exposure were statistically significant. The actual values of TTS fitted well with the values predicted with the TTS computational model. The results show that the computational TTS model gives results consistent with the TTS observed in the study group. Fitness instructors constitute a population at an increased risk of the hearing loss. Raising awareness of this fact and implementing hearing protection programs in this group of workers are urgently needed.

3:30

2pNSb3. 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.

3:45

2pNSb4. Evaluation of noise exposure and risk of hearing impairment in judicial transcribers. Kamil Zaborowski (Dept. of Physical Hazards, Nofer Inst. of Occupational Medicine, 8 Sw. Teresy Str., Lodz 91-348, Poland, kamil.zaborowski@imp.lodz.pl), Adam Dudarewicz, Malgorzata Zamojska-Daniszewska, and Malgorzata Pawlaczyk-Luszczynska (Dept. of Physical Hazards, Nofer Inst. of Occupational Medicine, Lodz, Poland)

The aim of this study was to evaluate noise exposure and risk of hearing impairment in judicial transcribers. Noise measurements and questionnaire inquiry were carried out in a group of 15 employees, aged 38.0 ± 0.8 years. Sound pressure levels emitted by headphones were measured using the MIRE technique according to ISO 11904-1:2008. The risk of noise-induced hearing loss was estimated according to the guidelines of ISO 1999:2013.The diffuse-field-related A-weighted equivalent-continuous sound pressure levels measured under headphones reached values from 62 to 80.7 dB. The study subjects worked as transcribers on average for 4–7 h a day. Consequently, the individual daily noise exposure levels in the examined group ranged from 61.2 to 79.9 dB. Such exposures for 10 years of occupational work pose a risk of hearing impairment (the mean hearing threshold at frequencies of 2, 3, and 4 kHz > 20 dB) reaching 3.5%. Almost three fourth of the employees reported their hearing deterioration (73.3%), whereas nearly a half of them experienced temporary hearing symptoms due to usage of headphones (43.3%).It is recommended that the transcribers should be covered by the hearing protection programme.

4:00

2pNSb5. Age difference of speech intelligibility while wearing hearing protective devices in noisy environment. Hiroyuki Hibino (Dept. of Health Policy and Management, Inst. of Industrial and Ecological Sci., Univ. of Occupational and Environ. Health, 1-8-219, Kitakyushu City, Fukuoka Prefecture 8070804, Japan, h-hibino@med.uoeh-u.ac.jp), Chikage Nagano, Natsuko Hoshuyama, Kahori Hashimoto, Kimie Fukuzawa, Kimiyo Mori, Jinro Inoue, and Seichi Horie (Dept. of Health Policy and Management, Inst. of Industrial and Ecological Sci., Univ. of Occupational and Environ. Health, Japan, Kitakyushu, Fukuoka, Japan)

Hearing conservation of elderly workers is important to keep their quality of life and to prevent occupational accidents. We performed this study to measure the age difference of hearing acuity in noisy environment. Subjects were total of 16 elderly (45 years old or more) and 21 healthy young people (35 years old or less). The two-syllable speech intelligibility with/without earplugs, with/without exposure to pink noises at 80 dB(A), 85 dB(A), and 90 dB(A) were measured in an anechoic room. For both age groups, the differences between the speech level required for 90% intelligibility and the noise level were nearly constant when the subjects wore ear plugs in noisy environment. Wearing ear plugs will be beneficial not only to protect hearing acuity but also to converse easily while wearing earplugs. However, special attention must be paid to talk loudly while wearing earplugs considering the Lombard effect.

4:15

2pNSb6. Mechanism of stiffness reduction in a thin broadband noise absorber. Lixi Huang (Mech. Eng., and Zhejiang Inst. of Res. and Innovation, The Univ. of Hong Kong, Haking Wong Bldg., Rm. 704, Hong Kong, Hong Kong, lixi@hku.hk)

Sound absorption by fibrous material is often considered broadband but poor in performance when the frequency is low. A new absorber is considered thin or otherwise when its thickness is compared with such a fibrous absorber for a given performance requirement. The performance of the latter is poor due to air stiffness in a compact design. This study first introduces a composite structure based on a parallel array of resonators with or without fibrous filling. The array is shown to introduce equivalent negative stiffness which is accompanied by a surge in system damping derived from resonator coupling. In principle, high damping is required for optimal absorption when the system as a whole is not operating at its resonance. However, the frequency dependency of such a composite system is too complex to be exactly optimal. The performance of the array structure is compared with a new fibrous absorber embedded with mass made from impervious membranes. The bandwidth and other characters of the two absorbers are compared when the incident sound has the typical environmental noise spectrum. Conclusions are shown to contradict with a few widespread misconceptions.

Session 2pPA

Physical Acoustics, Structural Acoustics and Vibration, and Computational Acoustics: Design of Acoustics Metamaterials: Optimization and Machine Learning II

Feruza Amirkulova, Chair
Mechanical Engineering, San Jose State University, 1 Washington Sq, San Jose, California 95152

Invited Papers

1:30

2pPA1. "Super-efficient gradient estimation technique," Recent advances in efficient adjoint sensitivity analysis and its application in metamaterial design. Laleh S. Kalantari (Comput. Sci., Univ. of Toronto, 661 University Ave. Ste. 710, Toronto, ON M5G 1M1, Canada, l.s.kalantary@gmail.com), Mohamed Bakr (Elec. and Comput. Eng., McMaster Univ., Hamilton, ON, Canada), and Marzyeh Ghassemi (Comput. Sci., Univ. of Toronto, Toronto, ON, Canada)

Computer-aided design (CAD) tools in electromagnetics allow accurate modeling of the preferred responses. We can adjust a desired response by conducting an optimization algorithm. This requisite the gradient estimation of structures with respect to potentially *N* number of optimizable parameters. In conventional gradient estimation methods, the number of required simulations scales linearly with *N*. Adjoint variable method (AVM) is an extremely efficient sensitivity analysis method that estimates the gradients with respect to the all *N* parameters by conducting only 2 simulations, regardless of *N*. We have developed AVM method for gradient analysis of anisotropic and dispersive anisotropic structures. Then, we applied it in wideband inversely invisibility cloak design for 2-D and 3-D structures at optical and microwave frequency regions. In those examples, our algorithm accelerates the gradient estimation over 1400 and 12 500 times per iteration, respectively, compared to the conventional methods. This advanced optimization based metamaterial cloak design to 3-D arbitrary shape objects and optical frequency region which was not feasible before. For the next step, we are planning to extend this efficient gradient estimation algorithm to the state of the art machine learning techniques and recently introduced continuous neural networks such as neural ordinary differential equations (ODE-Net).

1:50

2pPA2. Active surface sources for the exterior manipulation of acoustic fields. Daniel T. Onofrei (Dept. of Mathematics, Univ. of Houston, 4800 Calhoun Rd., 641 Philip G. Hoffman Hall, Houston, TX 77294-3008, donofrei79@gmail.com), Neil Egarguin, and Eric Platt (Mathematics, Univ. of Houston, Houston, TX)

In this talk, we will describe our recent results about the characterization of continuous boundary data (i.e., pressure or normal velocity) on active single sources or arrays for the approximation of different prescribed scalar wave field patterns in given exterior (near field or far field) regions of space. We will present the theoretical ideas behind our results as well as numerical simulations with applications in decoy characterization, field synthesis of personal audio spots and almost nonradiating sources with controllable near fields.

Contributed Papers

2:10

2pPA3. Optimization of one-dimensional Willis materials with hidden lumped elements. China M. Mauck (Dept. of Mathematics, Univ. of Utah, 125 S 900 E #15, Salt Lake City, UT 84102, mauck@math.utah.edu) and Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., Hanover, NH)

Willis coupling is a recently recognized material property that couples the pressure-strain and momentum-velocity equations and can lead to novel applications of metamaterials. We present an approach for optimizing the asymmetry factor (a non-dimensional measure of Willis coupling's effect on the specific acoustic impedance) in a metamaterial with lumped-element hidden degrees of freedom. The representative material element in the system we examine is a length of tube, which could be considered as a unit cell in a periodic system. Lumped-element features including side-branch resonators and membranes may be attached to the tube at various locations. We aim to determine optimal locations for each type of lumped-element feature, as well as optimal parameters in the design of the features themselves (e.g., for a resonator: volume, neck length, and neck cross-sectional surface area). An ideal design for a metamaterial exhibiting significant Willis coupling would result in a broadband non-negligible asymmetry factor.

2:25

2pPA4. Broadband suppression of total multiple scattering cross section using neural networks. Don Robert L. Pornaras (Mech. Eng., San Jose State Univ., 1 Washington Square, San Jose, CA 95192, donrobert.pornaras@sjsu.edu), Wei-Ching Wang (Mech. Eng., San Jose State Univ., San Jose, CA), Yanru Chen (Math, Foothill College, Los Altos Hills, CA), Grace Kwak (Pioneer High School, San Jose, CA), Feruza Amirkulova (Mech. Eng., San Jose State Univ., Springfield, MA), and Ehsan Khatami (Phys. and Astronomy, San Jose State Univ., San Jose, CA)

We will demonstrate a novel method to simulate acoustic multiple scattering by a configuration of cylinders and solve inverse problems using artificial neural networks (NN) and deep learning. We will research how to apply deep learning to solve inverse design problems efficiently. Solving

inverse design problems using optimization requires an iterative process of function evaluations and determining gradients, which are computationally expensive. In this work, forward multiple scattering problems are solved first by means of multiple scattering theory to provide training data for NN. Then, NN are trained to approximate the total scattering cross section (TSCS) function using backpropagation algorithm; the input of NN is positions of the cylinders, and the output is the TSCS evaluated at discrete values of wavenumber. Finally, trained NN are employed to solve inverse problems. Specifically, the TSCS by a plane configuration of cylinders is minimized over a range of wavenumbers using trained NN. A suppression of the TSCS can lead to the efficient design of broadband acoustic cloak. This method will be illustrated giving examples for a plane configuration of rigid cylinders.

2:40

2pPA5. Metaheuristic algorithm-based optimisation of an elastic metamaterial for robust control of multiple modes of vibration in a structure with parametric uncertainty. Lawrence Singleton (ISVR, Univ. of Southampton, Southampton SO17 1BJ, United Kingdom, ls2u17@soton.ac.uk), Jordan Cheer, and Stephen Daley (ISVR, Univ. of Southampton, Southampton, Hampshire, United Kingdom)

Undesirable resonant vibrations in a structure can be suppressed using various techniques. However, parametric uncertainty can reduce modelling accuracy and lead to a reduction in the efficacy of the designed suppression system. The suppression of multiple modes of vibration with robustness to parametric uncertainty is achievable using a large number of tuned-vibration-absorbers (TVAs) with distributed resonance frequencies. However, the performance of TVAs is also dependent on their positions relative to the mode shapes of the structure, which are not always known. Elastic metamaterials (EMMs) consist of distributed resonant substructures, at a scale which is small compared to the wavelength of vibration. This allows the material to be considered homogeneous with one set of defining physical properties, thus reducing the dependence on mode shapes. In this paper, a unit cell of a single-degree-of-freedom resonator-based metamaterial is defined, and the individual resonance frequencies are optimised using different metaheuristic algorithms for a nominal fixed-parameter beam and for the same structure with uncertainty in one or more of its parameters. The performance of the optimised metamaterial unit cell is then assessed when applied to more complex structures.

2:55

2pPA6. Design of acoustic metaclusters for the sound control using optimization and deep learning. Feruza Amirkulova (Mech. Eng., San Jose State Univ., 1215 Wilbraham Rd., Springfield, MA 01119, feruza.amirkulova@sjsu.edu), Don Robert L. Pornaras, and Wei-Ching Wang (Mech. Eng., San Jose State Univ., San Jose, CA)

We demonstrate a novel method to model finite metaclusters that can steer the energy of an incident wave preferentially toward a given direction. This is accomplished by solving an inverse multiple scattering problem for selecting a desired energy distribution of scattered waves. One can steer the incident energy toward a desired direction using a 2-D configuration of metacluster with finite number of fluid cylinders embedded in a homogeneous fluid medium. For a faster implementation of the method, we approximate the problem to small cylindrical particle limit which corresponds to low frequency scattering. The required mechanical properties of fluid scatterers are defined by T-matrix components obtained by solving linear system of equations. A major challenge in implementing our computational model and applying to design of metaclusters devices is ensuring that the scatterers remain manufacturable using available conventional materials with positive values of mechanical properties. Metaclusters are designed by minimizing the relative error between a given and computed scattering patterns and using advanced optimization algorithms and deep learning.

Session 2pPPa

Psychological and Physiological Acoustics and Speech Communication: Open Source Audio Processing Tools for Hearing Research I

Volker Hohmann, Cochair
Medical Physics, Universität Oldenburg, Postfach, Oldenburg 26111, Germany

Chaslav Pavlovic, Cochair
BatAndCat Corporation, 602 Hawthorne Avenue, Palo Alto, California 94301

Chair's Introduction—1:15

Invited Papers

1:20

2pPPa1. Usability assessment of a wearable speech-processing platform. Arthur Boothroyd (Speech, Lang., Hearing Sci., San Diego State Univ., 2550 Brant St., San Diego, CA 92101, aboothroyd@cox.net), Christine Kirsch, Carol Mackersie, Shaelyn Painter (Speech, Lang., Hearing Sci., San Diego State Univ., San Diego, CA), and Harinath Garudadri (Eng., Univ. of California, San Diego, La Jolla, CA)

The ultimate value of the NIDCD open-source initiative lies in use of the resulting platforms in research studies that advance knowledge and practice in hearing health-care. This study is concerned with usability of the current instantiation of the UCSD wearable speech-processing platform in possible field studies of advanced processing and/or listener-adjustment protocols. Usability is assessed, here, in the context of hearing-aid self-adjustment. Starting with an NAL-NL2 prescription for a relatively mild generic hearing loss, volunteers self-adjust gain and spectral tilt to preference in the clinic. They use the embedded version of the Boothroyd and Mackersie "Goldilocks" protocol while listening to recorded narrative at a conversational level. They are then accompanied by the researcher to local environments, with varying background noise and reverberation, and are given the opportunity to readjust. Listener reactions are subsequently obtained via a structured interview covering opinions on the physical attributes, acoustic attributes, and ease of adjustment. These are supplemented by coupler and real-ear measures of self-selected responses together with measures of phoneme recognition as a function of listening level. Individual differences are extreme but most participants respond positively to acoustics and usability and a proportion indicate a willingness to wear the platform in field studies.

1:35

2pPPa2. A high-fidelity multi-channel portable platform for development of novel algorithms for assistive listening wearables. Chaslav Pavlovic (BatAndCat Corp., 602 Hawthorne Ave., Palo Alto, CA 94301, chas@batandcat.com), Reza Kassayan, S. R. Prakash (BatAndCat Corp., Palo Alto, CA), Hendrik Kayser, Volker Hohmann (Univ. of Oldenburg, Oldenburg, Germany), and Andy Atamaniuk (EarLens Corp., Redwood City, CA)

The NIDCD has funded a number of projects to develop portable signal processing tools that enable real-time processing of the acoustic environment. The overarching goal is to provide a large group of researchers with the means to efficiently develop and evaluate novel signal processing schemes, individualized fitting procedures, and technical solutions and services for hearing apparatus such as hearing aids and assistive listening devices. We report here on a development done in the SBIR Phase II Project R44DC016247. This project builds on the software being concurrently developed in R01DC015429 to provide a complete portable and wearable software-hardware master hearing aid device needed for development of new solutions for assisted hearing. We will present and demonstrate the portable platform, currently in the Beta launch, that consists of a Cortex A8 based processing unit and a codec set able to support hearing aid architecture of up to 6 microphones and 4 speakers. It is currently accompanied by a binaural 2-microphone BTE hearing aid set, but will also support different headset form-factors of our partners. Additionally it features stereo line in and line out connections. The device can be remotely controlled with a smart phone.

1:50

2pPPa3. Real-time audio signal processing using system-on-chip field programmable gate arrays. Ross K. Snider (ECE Dept., Montana State Univ., 610 Cobleigh Hall, Bozeman, MT 59717, ross.snider@montana.edu), Trevor Vannoy, James Eaton, Matthew Blunt, E. Bailey Galacci, Justin Williams (ECE, Montana State Univ., Bozeman, MT), and Tyler B. Davis (Flat Earth, Inc., Bozeman, MT)

System-on-Chip (SoC) Field Programmable Gate Arrays (FPGAs) are ideal for real-time signal processing due to their low, deterministic latency and high performance. To showcase the utility of our open FPGA computational platform for real-time audio signal processing and computational modeling, several applications have been implemented. We have ported the openMHA hearing aid software [1] to our platform to show that pre-existing audio processing software can be implemented in SoC FPGAs by making external audio interfaces show up as a sound card. To highlight the ability to perform real-time computational modeling on our performance platform, we are implementing a real-time version of Laurel Carney's auditory-nerve model [2] running in its own custom accelerator in the FPGA fabric. To illustrate the ability to develop DSP algorithms in MathWork's Simulink and then implement them in the FPGA fabric we have taken several algorithms from Issa Panahi's group [3] to show both frame-based processing (noise reduction) and sample-based processing (dynamic range compression). Finally, we show that the platform can be used to visualize audio signals using a real-time spectrogram where FFTs are computed in the FPGA fabric. [1] www.openmha.org. [2] *JASA* **126**, 2390–2412. [3] www.utdallas.edu/ssprl/hearing-aid-project/.

2:05

2pPPa4. Open Master Hearing Aid (openMHA)—An integrated platform for hearing aid research. Hendrik Kayser (Medizinische Physik and Cluster of Excellence H4a, Carl von Ossietzky Universitaet, Ammerlaender Heerstrasse 114-118, Oldenburg D-26111, Germany, hendrik.kayser@uol.de), Tobias Herzke, Paul Maanen (HoerTech gGmbH, Oldenburg, Germany), Chaslav Pavlovic (BatAndCat Corp., Palo Alto, CA), and Volker Hohmann (Medizinische Physik and Cluster of Excellence H4a, Carl von Ossietzky Universitaet, Oldenburg, Germany)

The project R01DC015429 "Open community platform for hearing aid algorithm research" provides a software platform for real-time, low-latency audio signal processing: the open Master Hearing Aid (openMHA). It contains a versatile set of basic and advanced methods for hearing aid processing, as well as tools and manuals enabling the design of own setups for algorithm development and evaluation. Documentation is provided for different user levels, in particular for audiologists, application engineers and algorithm designers. The software runs on various computer systems including lab setups and portable setups. Portable setups are of particular interest for the evaluation of new methods in real-word scenarios. In addition to standard off-the-shelf hardware, a portable, integrated research platform for openMHA is provided in conjunction with the SBIR project R44DC016247. This contribution introduces openMHA and discusses the usage and possible application scenarios of the portable openMHA setup in hearing research. The opportunity is given to try a smartphone-based self-fitting application for the portable openMHA, and to learn about the flexible configuration and remote control of openMHA running a typical hearing aid processing chain. Furthermore, a discussion and exchange of ideas on current challenges and future developments is offered.

2:20

2pPPa5. Smartphone and algorithms: A platform for hearing study. Issa M. Panahi (Elec. and Comput. Eng., Univ. of Texas at Dallas, EC33, 800 West Campbell Rd., Richardson, TX 75080, issa.panahi@utdallas.edu) and Linda Thibodeau (Behavioral and Brain Sci., Univ. of Texas at Dallas, Richardson, TX)

Popular smartphones are portable devices with useful features such as powerful processor, microphones, speakers, low audio input/output delay, wire and wireless communication capability, graphical user interface (GUI) and easy-to-use touch screen. The smartphone features are improved every year. These features make smartphone a powerful stand-alone device for the development and real-time implementation of novel signal processing algorithms suitable for improving hearing study and hearing aid applications. In this paper, we present our smartphone-based platform and associated tools and documents for hearing study and hearing aid applications. An overview is given of the algorithms/apps developed to run on iOS and Android based smartphones in real time. Summary of the performance of developed algorithms/apps using smartphone is presented in comparison with other devices that are available in the market under different noisy conditions and low SNRs.

2:35

2pPPa6. On mitigating acoustic feedback in hearing aids with frequency warping by all-pass networks. Ching-Hua Lee (Elec. and Comput. Eng., Univ. of California, San Diego, Dept. of ECE, UCSD, 9500 Gilman Dr., La Jolla, CA 92093, chl438@eng.ucsd.edu), Kuan-Lin Chen, Fred Harris, Bhaskar D. Rao, and Harinath Garudadri (Elec. and Comput. Eng., Univ. of California, San Diego, La Jolla, CA)

Acoustic feedback due to the acoustic coupling between the microphones and loudspeakers at high gains can cause the hearing aid system to become unstable. This instability results in brief "howling" artifacts as magnitude and phase conditions fulfill the Nyquist stability criterion (NSC). We present a novel approach that uses all-pass networks to perform nonlinear spectral mapping that we call "freping," a portmanteau for frequency warping. In this contribution, we focus on spectral manipulations on top of adaptive feedback cancellation (AFC) to break NSC for improved feedback reduction. A real-time, multichannel realization is presented, which has individual control of the warping degree in each frequency band. Freping helps mitigate the NSC and thus leads to improved feedback control, while distortions due to freping are fairly benign based on informal subjective assessments. Our current findings indicate that improvements in terms of the perceptual evaluation of speech quality (PESQ) and the hearing-aid speech quality index (HASQI) can be achieved with freping for a basic AFC (PESQ: 2.56 to 3.52 and HASQI: 0.65 to 0.78) at medium amplification; and an advanced AFC (PESQ: 2.75 to 3.17 and HASQI: 0.66 to 0.73) at high amplification. Additional results will be included in the final presentation.

2:50

2pPPa7. SignalMaster Update 2019. Rafael E. Delgado (Intelligent Hearing Systems Corp., 6860 SW 81st St., Miami, FL 33143, redelgado@ihsys.com) and Patrick Davies (Intelligent Hearing Systems Corp., Miami, FL)

SignalMaster is a portable open speech and signal processing system that was developed specifically for researchers interested in developing their own real-time signal processing strategies. The system can be used for a wide range of research applications including development of hearing aid amplification and processing strategies and recording of evoked responses. SignalMaster uses a TMS320C6748 Digital Signal Processor (DSP) and provides two input and two output channels with 32 bit Analog-to-Digital (AD) and Digital-to-Analog (DA) convertors. Lithium-ion batteries provide up to 10 hours of freestanding operation. The system is available in two hardware platforms, the standard SignalMaster and the new SignalMasterLite, more compact and light weight for portable applications. The system may also be used while connected to a personal computer (PC) or Android Table via Bluetooth for additional control of experiments, off-line processing and data storage. A DLL provides functions for user developed applications to communicate with the hardware and be upload C language programs. This allows for different applications to be uploaded in order to reconfigure the processing algorithms being executed for any experiment at any time. DSP, PC-based and Android Table software source code examples provide users with the ability to develop their own user specific applications.

TUESDAY AFTERNOON, 3 DECEMBER 2019 — CROWN, 3:15 P.M. TO 4:45 P.M.

Session 2pPPb

Psychological and Physiological Acoustics and Speech Communication: Open Source Audio Processing Tools for Hearing Research II (Demonstrations)

Volker Hohmann, Cochair
Medical Physics, Universität Oldenburg, Postfach, Oldenburg 26111, Germany

Chaslav Pavlovic, Cochair
BatAndCat Corporation, 602 Hawthorne Avenue, Palo Alto, California 94301

Open hardware and software tools comprising basic and advanced features of commercial hearing aids are presented in their current state of development, including hardware to try on and software to interact with.

Session 2pSA

Structural Acoustics and Vibration: Characterization and Analysis of System Properties

Colby W. Cushing, Cochair
University of Texas at Austin, P.O. Box 8029, Austin, Texas 78751

Anthony L. Bonomo, Cochair
Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd, West Bethesda, Maryland 20817

Contributed Papers

1:15

2pSA1. Vibration characteristics of a car using transfer path analysis. Nihlatul Falasifah (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia) and Dhany Arifianto (Dept. of Eng. Phys., Institut Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id)

Vibration is a common issue in the automobile system, especially in an old car. Several causes that may generate vibration are vibration at idle and engine RPM. Vibration can decrease the ride comfort and become one of the contributing factors that cause an accident or emergency condition. Therefore, vibration evaluation to know the vibration characteristics of the classic car must be conducted to diagnose when and under what condition the vibration occurs. This paper aims to know the vibration level and vibration characterization of the classic car from car engine as the vibration source to some selected positions at low-frequency engine idle speed by using Transfer Path Analysis (TPA). This research used a 1991-manufactured car. Finite element calculation, simulation, and experimental modal analysis were used to determine the dynamic characteristics of the car for the natural frequencies. The natural frequencies that generated from the experimental modal analysis are convoluted by operational acceleration when the car engine was turning on which equal to operational force as transfer path analysis. In conclusion, TPA can be used to know the vibration characteristics in Noise Vibration and Harshness (NVH) issues by tracing the flow of vibration using operational force.

1:30

2pSA2. Pipe geometry calibration measurements for the improvement of ultrasonic clamp-on flow meters. Jack Massaad (Acoust. Wavefield Imaging, Delft Univ. of Technol., Lorentzweg 1, Delft 2628CJ, The Netherlands, J.M.MassaadMouawad@tudelft.nl), Douwe M. van Willigen (Electron. Instrumentation, Delft Univ. of Technol., Delft, The Netherlands), Paul L. van Neer (Acoust. and Sonar, TNO, The Hague, The Netherlands), Nico de Jong (Acoust. Wavefield Imaging, Delft Univ. of Technol., Rotterdam, The Netherlands), Michiel A. Pertijs (Electron. Instrumentation, Delft Univ. of Technol., Delft, The Netherlands), and Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., Delft, The Netherlands)

Clamp-on flow meters need a-priori knowledge of pipe geometry and material properties for their operation. Estimation of these properties may limit the accuracy of flow measurements. We are currently investigating the use of clamp-on flow meters based on matrix transducer arrays. Besides the benefits for flow measurement, these can be used for a-priori measurement of the required pipe properties. In the axial direction, average longitudinal wave speed and thickness of the pipe can be obtained from the dispersion curves of the Lamb waves in the pipe wall. In the circumferential direction, the arrival time of the fastest Lamb wave mode can be used to measure the pipe diameter. In our presentation, this method is discussed and proof of principle is provided through FEM simulations and measurements. The setup consisted of two pipes (outer diameters 64.5 mm and 93 mm), with two transducers (1 MHz) on top. The arrival time of the fastest Lamb wave mode was measured. For the outer diameter, the error between simulation and measurement was below 1% (small diameter) and below 3% (large diameter). The results suggest that further exploration of the unique possibilities of matrix transducers to accurately measure pipe parameters is opportune.

1:45

2pSA3. Application of transfer matrix method, Green's function, and variational technique to predict diffuse absorption with edge effect. Duyen Nguyen (Mech. and Aerosp. Eng., California State Univ., Long Beach, 1250 Bellflower Boulevard, Long Beach, CA 90840, DuyenThiMy.Nguyen@student.csulb.edu) and Allen Teagle-Hernandez (Mech. and Aerosp. Eng., California State Univ., Long Beach, Long Beach, CA)

Acoustic calculations utilizing the Transfer Matrix Method (TMM) occasionally underestimate the sound absorption performance of multi-layered porous material when compared to experimentally obtained data. This is particularly true when experimental data pertains to reverberant (Diffuse-Sabine) absorption. When there is a large impedance mismatch between the reverberant wall and the porous specimen, experimental data indicates Absorption Coefficients greater than 1, which defies the law of conservation of energy. It is hypothesized that the extra absorption is due to the diffraction of waves occurring at the edge of the tested materials, thus increasing the area where energy is absorbed. This contradiction of the law of energy is often referred to as the "edge effect". This study aims in creating a tool to predict and further understand the edge effect phenomena within reverberation and anechoic rooms and to determine the true absorption performance of the multi-layered porous material by using the TMM, Green's function, and the variational technique. A plate-foam system is used as a test bed; computation results are compared to experimental absorption data. The outcome of the research will support automobile, aerospace, and architectural companies in reaching sound and vibration targets required for new vehicle/building models.

2:00

2pSA4. Inverse acoustical characterization of porous material: A novel approach using diffuse field and transmission loss. Thiago Cavalheiro (Dept. of Mech. Eng., Universidade Federal de Santa Catarina, Laboratório de Vibrações e Acústica/UFSC, Florianópolis, Santa Catarina 88040900, Brazil, thiago.cavalheiro@lva.ufsc.br), Ricardo S. Rizzatti (Dept. of Mech. Eng., Universidade Federal de Santa Catarina, Florianópolis, Santa Catarina, Brazil), Fabio Luis V. Kulakauskas (Vibtech, Arujá, São Paulo, Brazil), Lucas V. Kulakauskas, Luisa P. Serafim, and Arcanjo Lenzi (Dept. of Mech. Eng., Universidade Federal de Santa Catarina, Florianópolis, SC, Brazil)

Porous materials are largely used to improve sound quality in enclosed spaces. Inverse acoustical characterization of porous media is becoming

popular due to experimental practicality. Most of the studies have as input data some impedance tube measurements. However, this approach presents several complications, mainly concerning the sample boundary conditions. The purpose of this study is to obtain the macroscopic parameters by inverse characterization, using transmission loss and the absorption coefficient as input data, both measured in reverberant chamber. One-piece and partitioned 2 m^2 samples were measured. Several low-density materials were analyzed. The influence of boundary conditions, acoustic diffuse field and impervious screen layering porous material have been investigated. The Transfer Matrix Method is combined with the Johnson-Champoux-Allard equivalent fluid model on an optimization process. Transmission loss and absorption coefficient are simultaneously calculated. The mean square error between calculated and measured values is set as the objective function over the frequency range. Although the mounting condition of the measured transmission loss setup has required appropriate adjustment, good satisfactory repeatability on the optimized parameters was achieved.

2:15

2pSA5. Investigations into the accuracy of the light damping approximation. Alyssa T. Liem (Boston Univ., 110 Cummington Mall, Boston, MA 02215, atliem@bu.edu) and James G. McDaniel (Boston Univ., Boston, MA)

This presentation surveys a variety of representative case studies in an effort to better understand and predict the accuracy of the light damping approximation. For viscously damped systems, the light damping approximation ignores the off-diagonal elements of the transformed damping matrix. When the approximation is valid, it presents profound advantages for numerical simulations. The most obvious of these is uncoupling of the equations of motion. A less obvious advantage is the ability to estimate the viscous damping matrix from a knowledge of the modal loss factors and undamped eigenvectors. The presentation begins by defining error metrics for quantifying the accuracy of the approximation in time and frequency domains. Next, case studies are designed to understand how statistical properties of the structure control the accuracy of the approximation. These investigations make two fundamental contributions. The first contribution is the knowledge required to estimate the accuracy of the approximation without significant computational expense. The second contribution is identification of statistical properties that are important to the construction of model libraries used in training machine learning algorithms.

2:30–2:45 Break

2:45

2pSA6. Dynamic mechanical characterization of viscoelastic materials using bending wave correlation vibrometry. Max Miller (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St, Troy, NY 12180, millem23@rpi.edu) and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Direct solid-borne sound measurements allow incorporation of phase in characterizing viscoelastic materials. In the absence of excessively large samples, reflections from specimen boundaries obscure this valuable data. Circularly crested bending waves generated in plates provide a means of preserving phase information while maintaining reasonable sample sizes. The coherence of signals measured equidistant from the source and the lack thereof concerning reflections, enables isolation of the direct sound. This talk presents developments in a laser Doppler vibrometer based dynamic mechanical characterization method aimed at broadband classification of viscoelastic materials. The applicability of this technique to *in-situ* applications is demonstrated. Material properties are shown to correspond with results from existing methods.

3:00

2pSA7. Bounds on the imaginary part of complex Poisson's ratio of viscoelastic materials. Tamás Pritz (Budapest Univ. of Technol. and Economics, Műegyetem rkp. 3-9, Budapest 1111, Hungary, tampri@eik.bme.hu)

Bounds on the imaginary part of complex Poisson's ratio of viscoelastic materials The Poisson's ratio of ideally elastic solids is a real number for either static or dynamic loading, and has bounds, namely its magnitude can be between -1 and ½ in case of homogeneous, isotropic, linear materials. In contrast, the Poisson's ratio of viscoelastic solids subjected to dynamic loading can be considered as a complex number, referred to as complex Poisson's ratio (CPR), because of the material damping. It is clear that the real part of CPR has the same bounds as the Poisson's ratio. The question what are the bounds on the imaginary part of CPR is investigated in this paper for homogeneous, isotropic, linear viscoelastic materials with positive Poisson's ratio. It is shown that the imaginary part of CPR also has bounds, which depend on the Poisson's ratio itself and the material damping. Equations are developed to determine the bounds in question as functions of the Poisson's ratio and shear loss factor provided that the latter is lower than 0.3. It is proved that the imaginary part of CPR cannot be larger than approximately one tenth part of the shear loss factor. Experimental data supporting the theoretical findings are presented.

3:15

2pSA8. Structural transients from the impact force excitation of beams. Peter Stepanishen (Dept of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02881, steppipr@uri.edu)

Analytical and experimental results are presented for the transient longitudinal and flexural waves which result from the impact force excitation of beams. Space-time, integral transform and convolution methods are utilized for the analyses. Analytical results are presented for the *in vacuo* velocity response of an infinite, semi infinite and a beam of finite length to illustrate the effects of boundary reflection and material dissipation on flexural and longitudinal waves. Experimental results are then presented to illustrate the general behavior of transient longitudinal and flexural wave propagation in a free-free beam of finite length *in vacuo* which is impacted by a hammer excitation. In contrast to the clearly observable multi-pulse structure for the reflected longitudinal waves the dispersive nature of the flexural waves results in overlapping reflected waves with a loss of the multi-pulse structure. Frequency-time analyses provide a clearer picture of the underlying physics and illustrate the expected dispersion relationships.

3:30

2pSA9. Acoustic transients from the impact force excitation of beams and wind chimes. Peter Stepanishen (Dept of Ocean Eng., Univ. of Rhode Island, Narragansett, RI 02881, steppipr@uri.edu)

A time dependent modal based approach is presented to address the structural acoustic transients resulting from the impact excitation of a beam or wind chime. The normal velocity of the structural element is first expressed as a space-time modal expansion using the *in-vacuo* normal modes as basis functions. A convolution integral equation approach is presented to determine the time dependent modal velocities where fluid loading is included via the use of self and mutual modal radiation impulse responses which provide modal coupling. The modal radiation impedances and impulse responses are obtained from the associated radiation resistances via the use of Hilbert and Fourier transform relationships. Time-dependent dipole modal distributions along the beam or wind chime are then used to determine the pressure field using a convolution integral approach with the modal velocities and modal space-time impulse responses which are simply related to the *in-vacuo* modes . Numerical results are presented to illustrate interesting coincidence effects in the transient acoustic far field from beams and wind chimes.

Session 2pSP

Signal Processing in Acoustics, Animal Bioacoustics, Underwater Acoustics, Acoustical Oceanography, and Speech Communication: Signal Processing for Biological Transients

Simon E. Freeman, Cochair
Naval Undersea Warfare Ctr., 7038 Old Brentford Road, Alexandria, Virginia

Blaine M. Harker, Cochair
Naval Undersea Warfare Center Division Newport, 1176 Howell St., Newport, Rhode Island 02841

Philip Caspers, Cochair
Naval Undersea Warfare Ctr., 1176 Howell St., Newport, Rhode Island 02841

Chair's Introduction—1:00

Invited Papers

1:05

2pSP1. Formation of the broadband biosonar beam of dolphins: Melon-focusing hypothesis not correct. Whitlow Au (Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, wau@hawaii.edu) and Chong Wei (Ctr. for Marine Sci. & Technol., Curtin Univ., Singapore, Singapore)

In 1974, Norris and Harvey measured the sound velocity profile of the melon of a just deceased bottlenose dolphin and found a low-velocity core in the melon with the velocity increasing towards the surface. This was the genesis of the "melon focusing" hypothesis in the formation of the biosonar beam. Aroyan solved the wave equation for a signal produced at the nasal plug using a finite difference approach and found that the skull was responsible for the biosonar beam. Aroyan's results were never taken seriously and the notion that the melon was mainly responsible for the biosonar beam continues to this day. We used a finite element approach to solve the wave equation for a broadband signal generated at the phonic lips. CT scan data were used to provide a map of the acoustic impedance of the dolphin's head and our results compared well with different measurements made around the dolphin's head and in the far field. Our findings also showed that the skull and air sacs are the most influential structures in the formation of the biosonar beam and the melon was responsible for directing the axis of the beam. Our results should put to rest the "melon-focusing" hypothesis.

1:25

2pSP2. Hydrophone array processing of biological transient sounds in an acoustically complex environment. Simon E. Freeman (Sensors and Sonar Systems, Naval Undersea Warfare Ctr., 7038 Old Brentford Rd., Alexandria, VA 22310, simon.freeman@gmail.com), Lauren Freeman (Naval Undersea Warfare Ctr., Newport, RI), Radienxe Bautista (Sensors and Sonar Systems, Naval Undersea Warfare Ctr., Newport, RI), Philip Caspers, Blaine M. Harker, Alexis Johnson (Naval Undersea Warfare Ctr., Newport, RI), and Chris Toole

Biological transients from invertebrates and fishes dominate the coral reef sound field, an acoustically and biologically complex ecosystem that is challenging to monitor effectively. Three geometries of a reconfigurable mid-frequency hydrophone array were used to quantify the reef biological sound field and investigate bioacoustic changes associated with organism interactions and environmental variables stemming from soniferous reef organism activity. In the three cases, array geometry was optimized to maximize resolution in the horizontal, provide equal vertical and horizontal aperture, and to provide optimal two-dimensional resolution without spatial ambiguity over the reef surface. Reconstructed time-series from beamformer outputs were processed through a number of techniques. Simultaneous underwater video from static cameras enabled validation of reef activity during daylight hours.

1:45

2pSP3. Convolutional generative models for underwater acoustic event classification. Paul M. Baggenstoss (FKIE, Fraunhofer, Fraunhoferstr. 20, Wachtberg 53474, Germany, p.m.baggenstoss@ieee.org)

Despite the success of discriminative (DISC) classifiers, there remain serious flaws in the DISC approaches. Using adversarial sampling, a DISC network can be fooled into producing any desired classifier output by subtle changes in the data sample. The problem lies in the goal of DISC methods: assigning class identity without further considerations. In contrast, generative (GEN) methods model the underlying data generation process, can be interrogated and groomed by researchers, so have the potential to operate well with little training data. But, the GEN task is more difficult, so performance lags behind. But, given time and effort, GEN performance can even

surpass performance of DISC methods as shown by deep belief network of Hinton in 2006 which performed better than comparable fully-connected (non-CONV) DISC networks. As CONV DISC networks (CNNs) have seen a quantum leap in performance, GEN methods have once again fallen behind. This talk details unexplored avenues to greatly improve CONV GEN models (CGMs). One avenue, "max-pooling for CGMs" is backed up by promising experiments. Max-pooling is a dimension-reduction step that is partly responsible for the success of CNNs, but presents a severe obstacle for CGMs by discarding the pooling positioning information (PPI). We show that encoding PPI info the features greatly improves the quality of GEN auto-encoders. When PPI together with added GEN neurons are "appended" to existing DISC CNNs, a hybrid GEN/DISC CNN is created with the best qualities of both approaches.

2:05

2pSP4. Quantitative evaluation of standard and enhanced feature extraction from individual biological sounds recorded in coral reef environments. Kelley McBride (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093, kmcbride@ucsd.edu), Gerald L. D'Spain (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), David K. Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ., Newport, OR), Alison B. Laferriere, Mike Nicoletti (Raytheon BBN Technologies, Cambridge, MA), and Michael J. Buckingham (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Passive acoustic monitoring of three protected coral reef habitats south of St. John Island in the US Virgin Islands was conducted in April, 2019. Four wideband Autonomous Passive Acoustic Monitoring (APAM) packages, each with 12-element hydrophone arrays, were deployed on the 10-m-deep seafloor and recorded continuously over the 9-day experiment. Previously published methods for robust feature extraction of individual biological sounds [Mellinger and Bradbury, 2007] have been applied to these data and compared with enhanced approaches to feature extraction. This talk presents results of quantitative evaluations of the performance of these methods. Feature extraction is categorized as a parameter estimation problem whose performance is quantified by the statistical distribution of estimates, typically summarized by the bias and variance of these distributions. Results for a simulated case first are presented, and then those from the APAM-recorded data. The enhanced approaches show significant improvement over the standard approaches for several of the extracted features.

2:25

2pSP5. Signal and array processing of biological transients. Gerald L. D'Spain (Marine Physical Lab, Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu), Goldie T. Phillips (Sci-Brid Int. Consulting, LLC, Arouca, Trinidad and Tobago), and Dennis Rimington (Marine Physical Lab, Scripps Inst. of Oceanogr., San Diego, CA)

Studies of biological transients in the time domain require removing the distorting effects of the data acquisition system (DAQ) system. Unfortunately, amplitude and phase calibration across frequency (equivalently, the DAQ impulse response) rarely is determined for bioacoustic systems. Calibration is not needed for certain types of processing, e.g., detection since signal and noise undergo the same distortion. Under certain assumptions (mainly pertaining to the properties of the noise), the optimal detector for transients is a member of the family of power-law detectors. Therefore, a challenge is determining the pre-processing approach that converts ocean acoustic noise into noise with the assumed properties. A direction-of-arrival estimation approach that is optimal under certain conditions is the minimum variance, distortionless response beamformer. However, transients typically are snapshot-deficient for estimating the data cross spectral matrix. Quantitative performance of sub-optimal but robust approaches with passive acoustic arrays deployed in a variety of ocean environments are investigated. A non-acoustic biological transient is the infrared radiation from the heat of a marine mammal blow. Results from an automated processor based on the power-law detector for infrared video are presented.[Support from the Office of Naval Research, the Joint Industry Programme, and the Walton Foundation].

2:45

2pSP6. Signal processing for biological transients emitted by dolphins and snapping shrimp. Brian G. Ferguson (Acoust. Systems, Defence Sci. and Technol., Locked Bag 7005, Liverpool, New South Wales 1871, Australia, Brian.Ferguson@defence.gov.au) and Eric L. Ferguson (School of Elec. and Information Eng., The Univ. of Sydney, Sydney, New South Wales, Australia)

In temperate and tropical waters, the dominant source of biological noise in shallow bays, harbors and inlets is snapping shrimp, which can adversely affect the performance of high-frequency sonar systems. The modified method for passive ranging by wavefront curvature enables the source position and source level of individual snaps to be measured *in situ*. The results are presented for a sequence of 1000 snaps. The principal habitat of the local snapping shrimp population coincides with a narrow wharf, which is 120 m in length. Next, this passive ranging method is used to localize free-ranging dolphins in their natural habitat. The peak-to-peak sound pressure levels of the biosonar pulses (clicks) emitted by Indo-Pacific dolphins (*Tursiops aduncus*) are measured for source ranges from 30 m to 300 m, even when the dolphins are echolocating at the same time. The waveforms and signal properties of the click signals are presented, along with their effectiveness for probing the shallow-water environment. The bistatic scattering impulse responses of the sea floor, sea surface and scatterers in the water volume are shown. Finally, consideration is given to the contexts in which snaps and clicks are made.

3:05–3:20 Break

3:20

2pSP7. How DIFAR sensors can enhance detection and 2-D localization of impulsive fish sounds on coral reefs. Aaron Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr, MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu), Alexander Conrad (Greeneridge Sci., Santa Barbara, CA), Ludovic Tenorio-Hallé (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), Simon E. Freeman (Naval Undersea Warfare Ctr., Alexandria, VA), Lauren Freeman (Naval Undersea Warfare Ctr., Washington, DC), and Katherine H. Kim (Greeneridge Sci., Santa Barbara, CA)

The impulsive sounds produced by tropical fish are a prominent component of coral reef acoustic environments off Hawaii. The resultant ambient noise field is highly nonstationary, making it difficult to equalize the noise background when implementing standard intensity-based detectors on conventional hydrophones. Here we demonstrate how DIFAR sensors can be used to enhance the contrast between transient fish signals and background ambient noise, permitting simultaneous detection and triangulation of individual pulses. This approach assigns an azimuth to each time-frequency component of a conventional spectrogram, by computing the arctangent of the active intensity measured on two orthogonal axes. The resulting "azigram" can be processed using standard image processing methods to isolate connected regions that share the same azimuth, and to match similar regions on azigrams from nearby DIFAR sensors. The cross-matched bearings can then be used to triangulate the source. The technique is being used to study "hotspots" of fish activity on coral reef pinnacles.

Contributed Papers

3:40

2pSP8. Detecting shifts in coral reef soundscape with unsupervised learning. Emma Reeves Ozanich (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, ecreeves@ucsd.edu), Peter Gerstoft (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Chris Toole, Lauren Freeman, Simon Freeman, and Alexis Johnson (Naval Underwater Warfare Ctr., Newport, RI)

Biological sources contribute significantly to coral reef ambient noise environments, yet the ecosystem-level mechanisms of temporal and spatial variation in the reef soundscape are not well understood. In this study, subtle shifts in reef ambient noise are examined using unsupervised machine learning on hydrophone array data. Unsupervised learning does not require data labels, but uses nonlinear inference to find explanatory features within the data. A hydrophone array was used to generate spatially filtered time series inputs for machine learning. Video cameras were collocated and time-synced with the hydrophone array to provide nominal ground-truth. We discuss the tradeoff parameters of the unsupervised learning methods. Changes in the dominant data features during the experiment are compared to the video recordings and researcher observations.

3:55

2pSP9. Analysis of spatial-temporal variations in coral reef transients observed off the coast of Hawaii. Philip Caspers (Naval Undersea Warfare Ctr., 1170 Howell St., Newport, RI 02841, philip.b.caspers@navy.mil), Blaine M. Harker, Chris Toole, Simon Freeman, and Lauren Freeman (Naval Undersea Warfare Ctr., Newport, RI)

Short duration acoustic transients are a conspicuous mechanism of sound generation by organisms within a coral reef environment. In this work, we investigate patterns of transients recorded in a coral reef environment during a crepuscular period associated with interactions between species. Video synchronized to hydrophone array element recordings at three different locations were used to identify times and positions of coral reef species interactions, i.e., predator-prey interactions. Normalized, directional audio reconstructions along fixed bearing directions were analyzed with a power law detector at different temporal resolutions to identify the occurrence of transient events. Segmented transients were collated with time and incident direction information and grouped into temporal windows within the audio reconstruction. Dimensional reduction of the transient audio was applied and changes in temporal characteristics of the compressed features associated with particular species were evaluated. Analysis of the order-statistic normalizer and power law detector parameters to detect biological transients is also discussed.

4:10

2pSP10. Discrimination of chronic and transient sound sources in marine soundscapes. Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr, La Jolla, CA 92093, sbaumann@ucsd.edu), Kaitlin E. Frasier (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., San Diego, CA), Megan F. McKenna, Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO), Jenni Stanley (Northeast Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, Woods Hole, MA), Tetyana Margolina, John E. Joseph (Oceanogr., Naval Postgrad. School, Monterey, CA), and Leila Hatch (Stellwagen Bank National Marine Sanctuary, National Oceanic and Atmospheric Administration, Scituate, MA)

Soundscapes are comprised of chronic and transient sounds that overlap each other in time and frequency. These sound sources can be of anthropogenic, geological or biological origin. We automatically discriminated and classified sources with an unsupervised learning strategy based on clustering and subsequent training of a neural net. The parameter space was comprised spectral features and variation of these features over time. We document how different time and frequency binning influenced the outcome of the machine learning for the different sources. This analysis is based on multi-month data collected within several US National Marine Sanctuaries as part of the SanctSound project (https://sanctuaries.noaa.gov/science/monitoring/sound/). The project aims to characterize soundscapes within sanctuaries including assessing the influence of anthropogenic noise on marine life in their boundaries. We are testing whether the automated process helps to gain comparable results across the very different sanctuary habitats located in temperate to subtropical waters.

4:25

2pSP11. Improvements in detection, localization, and tracking of sperm whales (*Physeter macrocephalus*) in Kauai. Gabriela C. Alongi (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, gabriela.alongi@nmmf.org), Stephen W. Martin, Brian Matsuyama (National Marine Mammal Foundation, San Diego, CA), Cameron R. Martin, Roanne Manzano-Roth (Naval Information Warfare Ctr. Pacific, San Diego, CA), and Elizabeth Henderson (Naval Information Warfare Ctr. Pacific, San Diego, CA)

Since 2002, passive acoustic monitoring at the Pacific Missile Range Facility off Kauai, HI has yielded minimum density estimates and disturbance analyses for various whale species, but the ability to do so for sperm whales (*Physeter macrocephalus*) has been limited. Such efforts have traditionally

2p TUE. PM

suffered from false positives and decreased localization accuracy with increased click density. Recent software developments have aimed to address these issues and attempts were made to quantitatively and qualitatively assess improvement using archived 2014 data (n = 1063 hours of recording). Algorithm changes generated 2.4 times more detections (SD = 3.9x) and 1.4 times more localizations (SD = 2.4x) on average. Metrics indicating localization accuracy also improved. The number of detections per localization increased from a median of 5.5 (IQR = 5.2–5.8) to 7.0 (IQR = 6.0–8.3) and the percent of localizations theoretically capable of being tracked increased from 5.9% (IQR = 1.4%–11.6%) to 31.2% (IQR = 8.0%–46.9%). Visual comparison of localizations indicates much lower false positive rates, while other software developments have enabled tracking slow clickers and identifying foraging groups. Ultimately, track-level analyses should permit a more stable metric for minimum density estimates and disturbance tests and help establish baseline kinematics and behavior.

4:40

2pSP12. Differentiating marine mammal clicks using time-series properties. Bruce Martin (JASCO Appl. Sci., 32 Troop Ave., Ste. 202, Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com), Xavier Mouy (JASCO Appl. Sci., Victoria, BC, Canada), Briand Gaudet, and Katie Kowarski (JASCO Appl. Sci., Dartmouth, NS, Canada)

Automatic detection of marine animal vocalizations is increasingly used to analyse the extensive acoustic datasets collected from autonomous passive acoustic recorders, resulting in a constant effort to improve detector accuracy and develop new and more efficient detection methods. Differentiating between the clicks produced by odontocete species can be especially problematic due to overlapping time and frequency characteristics. Classifiers for clicks are often based on short-time Fourier transforms or Wigner-Ville transforms which are computationally expensive. We propose a computationally efficient method of detecting and differentiating between clicks based on a two stage classifier. First, an initial classification is obtained using three features derived from the time series of the click. Second, the cepstrum is used to determine the inter-click interval. The method was tested on recordings which included clicks produced by small dolphins, killer, pilot and sperm whales, as well as at least three species of beaked whales. This new approach increases the efficiency of analysis and provides reliable species classification.

4:55

2pSP13. Processing seismic source transients to quantify ecosystem impact. Natalia Sidorovskaia (Phys., UL Lafayette, UL BOX 44210, Lafayette, LA 70504-4210, nas@louisiana.edu) and Kun Li (Phys., UL Lafayette, Lafayette, LA)

Modeling of acoustic field metrics for transient seismic signals in ocean environments is required for the regulatory and resource management purposes. The validation by calibrated measurements is a critical step to assess model's accuracy. The talk discusses acoustic metrics derived from the field data collected in the Northern Gulf of Mexico. The purpose of the experiment was to characterize 3-dimemsional field of the industrial seismic exploration array. Received Sound Pressure Levels of direct arrivals showed a large variability (up to 50 dB re 1 μPa) for a fixed distance between source and receiver indicating that the distance cannot be used as a single parameter to derive meaningful exposure thresholds. However, the far-field primary acoustic field variations with distance along the true acoustic path for a narrow angular bin are accurately predicted using a simplified model of the theoretical monopole source in free space. The source level of a monopole depends on radiation direction. The different acoustic metrics show robust relationships. As distance between source and receiver increases, the direct arrival may not be present (shadow zone). In this case more comprehensive propagation modeling is required. [Research funded by the Sound and Marine Life Joint Industry Programme.]

5:10

2pSP14. Graph signal smoothness for direction-of-arrival estimation of source targets using non-uniform line arrays. Eldridge Alcantara (Elec. & Comput. Eng., Univ. of Washington, Seattle, Box 352500, Seattle, WA 98195, eealcant@uw.edu), Les Atlas (Elec. & Comput. Eng., Univ. of Washington, Seattle, Seattle, WA), and Shima Abadi (Mech. Eng., Univ. of Washington, Bothell, Bothell, WA)

Performing direction-of-arrival (DOA) estimation of a detected source target using data collected from an array of sensors is a long-studied problem in signal processing. The emerging framework of signal processing on graphs (SPG) offers the opportunity to look at this problem from a different point of view by adding the concept of a graph underlying the measured data as an extra layer of information that can be incorporated in to the data processing system. While a recent presentation [*J. Acoust. Soc. Am.* **143**, 1852 (2018)] showed how to use the Graph Fourier Transform to estimate DOA on a uniform line array, the purpose of this talk is to show how we can use the smoothness of a graph signal across vertices for the more involved case of a non-uniform line array. The scope of the problem was narrowed to DOA estimation of a single, far-field, and narrowband source signal. An iterative approach was developed that demonstrates the utility of SPG, and the approach was verified and tested through simulation under different signal-to-noise ratios. We also compared the results to conventional delay-and-sum beamforming to gain insight on what differentiates SPG over existing methods.

5:25

2pSP15. A program-dependent digital compressor for auditory prostheses specialized for signal transients. Eric W. Tarr (Audio Eng. Technol., Belmont Univ., 1900 Belmont Blvd., Nashville, TN 37212, eric.tarr@belmont.edu)

Dynamic range compressors found in hearing aids and cochlear implants are designed to have a programmable response time. In particular, the attack and release parameters control the rate at which a compressor responds to the onset and offset of a signal transient. Some previous perceptual studies have found preferences for faster response times, while other studies have found preferences for slower response times. An explanation for the conflicting results is that the experiments' stimuli and perceptual measures have varied. These results are indicative of the challenge in designing compressors for a wide range of speech, environmental, and musical signals each with unique transient characteristics. Researchers have proposed using multiple wide-band compressors or parallel narrow-band compressors with different response times. This research presents an alternative approach for a single compressor having adaptive, program-dependent, attack and release times. Specifically, the response times depend on the rate of change of the compressor's gain reduction. When a transient causes a large change in gain reduction, the response times are faster. During periods with small changes in gain reduction, the response times are slower. Considerations for real-time implementation are discussed.

Session 2pUW

Underwater Acoustics: Propagation and Sound Generation Physics and Modeling

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Contributed Papers

1:00

2pUW1. Effective representation of sound propagation in ocean using Hartree-Fock Bogoliubov theory. David P. Knobles (Knobles Sci. and Anal., PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com)

Physical insight into the complexities of sound propagation in the ocean can be achieved by considering effective waveguides; for example, Zhao *et al*., "Modeling of Green's function with bottom reflective parameters (P,Q) instead of Geoacoustic parameters," *J. C. A.* **22**(1), 1440005, (2014).). In principle, effective waveguide representations can enhance the efficacy of solving the statistical inverse problem. Inspired by such work, an attempt is made to utilize methods to solve the quantum many-body problem. One way to solve the nuclear many-body problem (NMBP) is to transform a group of strongly interacting particles into a set of weakly interacting particles within a potential that results from the collective nature of the two-body interactions. The most noteworthy of these approaches is the Hartree-Fock-Bogoliubov (HFB) theory. In analogy with the NMBP we replace an ocean acoustics problem that is described by a set of strongly coupled modes with a self-consistent modal potential that contains all the long-range quasi-mode correlations. Computational results are provided that compares the standard coupled mode solution to the HFB generated solution.

1:15

2pUW2. Characterization of sound waves propagating in a fluctuating ocean: experimental validation with ALMA. Gaultier Real (DGA Techniques Navales, Ave. de la tour royale, Toulon 83100, France, gaultier.real@gmail.com) and Dominique Fattaccioli (DGA Techniques Navales, Toulon, France)

The authors present an acoustic system, designed by DGA Naval Systems, dedicated to the study of sound propagation in challenging environments. The system is called *ALMA* for *Acoustic Laboratory for Marine Applications*. Shallow and coastal waters, where oceanographic phenomena (such as linear internal waves, tides and 3-D effects) interact with acoustic propagation, represent the main area of deployments. Since 2014, 5 at-sea campaign have been successfully conducted in the Mediterranean Sea and in the Atlantic Ocean. They mainly consisted in propagating sound waves in the 1–15 kHz frequency band using fixed or towed sources towards a modular passive acoustic array. The latter is composed of 8 rigid arms carrying 16 hydrophones each. These arms can be, and actually were, arranged in various shapes, depending on the goal of the experiment. Environmental sensing using thermistor strings and CTD cast complete the experimental equipment. The analysis of the 2016 campaign demonstrates the ability of the system to gather data representative what was described as the "saturation" theory by S. Flatté in the 1980s. In fact, criteria based on the mutual coherence function and the normalized acoustic intensity probability density function allow to explain the observed variability of detection performance on a 4x32-hydrophones vertical comblike passive acoustic array. A global progress report on the use of data gathered by the ALMA system, as well as future plans—including deployment in high latitudes environments—will be discussed.

1:30

2pUW3. Mode coupling and scattering in a submarine canyon environment. Brendan J. DeCourcy (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., Iselin 3 (MS# 55), Falmouth, MA 02543, bdecourcy@whoi.edu) and Timothy F. Duda (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Acoustic propagation in a realistic ocean environment based on the Hudson Canyon is computed using an Omnidirectional Coupled Mode (OCM) and a 3-D Parabolic Equation (3DPE) approach. The 3DPE is limited by nature to one-way wave radiation, while the OCM is not. The mode coupling method accounts for omnidirectional propagation and refraction, while the forward-propagation PE model can more efficiently approximate the canyon acoustics while ignoring some of the complicated refraction. The primary motivation is to explore the differences in output between the two methods in pursuit of describing the relative importance of environmental uncertainty in model inputs. Examples of features of interest include steep bathymetric slopes, and sound speed fronts in the water column. Some methods for adapting the computational approach in the OCM model based on environmental properties are outlined, and some sensitivity analysis is presented. [Work supported by ONR.]

1:45

2pUW4. Underwater sound propagation modeling and the predictive probability of detection framework. Timothy F. Duda (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu)

The predictive probability of detection framework (PPD) provides a measure of uncertainty to the binary problem of signal detection in the situation of fluctuating received signal. The fluctuations can be from many causes, including short-term and difficult to predict propagation variability, long-term and possible to predict propagation variability, and noise variability. In PPD, the density functions covering all fluctuating-causing processes are combined and analyzed with respect to a benchmark to give a detection probability. In the traditional PPD analysis, the benchmark propagation loss is a somewhat general function of range, as are other quantities such as array gain. These provide reference levels for signal to noise ratio evaluation. With more detailed data-informed propagation modeling, including 3-D modeling, additional structure can be given to those functions, and the stochastic fluctuation aspects of the method can potentially be reduced, thereby altering computed maps of probability of detection. Example computations will be shown for spatially heterogeneous environments: canyons, slopes, and internal wave areas. [Work supported by the Office of Naval Research.]

2:00

2pUW5. 3-D acoustic propagation through an estuarine salt wedge. D. Benjamin Reeder (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943, dbreeder@nps.edu) and Ying-Tsong Lin (AOPE, Woods Hole Oceanographic Inst., Woods Hole, MA)

The estuarine salt wedge presents a dynamic and highly refractive waveguide, the acoustic propagation characteristics of which are controlled by the water column sound speed gradient and boundary interactions. Acoustically, the salt wedge consists of two isospeed layers separated by a thin, three-dimensional, high-gradient layer. The behavior of a broadband (500–2000 Hz) acoustic field under the influence of an estuarine salt wedge in the Columbia River estuary is explored using two 3-D acoustic propagation models: 3-D rays and 3-D parabolic equation (3DPE). These model results are compared to data collected during a field experiment in 2013. Results demonstrate that the dominant physical mechanism controlling acoustic propagation in this waveguide shifts from 3-D bottom scatter in a non-refractive waveguide (before the entrance of the salt wedge) to 3-D acoustic refraction with minimal bottom interaction in a refractive waveguide (when the salt wedge occupies the acoustic transect). Vertical and horizontal refraction in the water column and out-of-plane scattering by the bottom are clearly evident at specific narrowband frequencies; however, these mechanisms contribute to, but do not account for the total observed transmission loss.

2:15

2pUW6. Seismo-acoustic wave propagation in the Rade of Hyères (France) generated by counter-mining of explosive devices: comparison between numerical simulations and real experiments. Fang Wang, Nathalie Favretto-Cristini (Aix Marseille Univ., CNRS, Centrale Marseille, LMA, Marseille, France), Paul Cristini (Aix Marseille Univ., CNRS, Centrale Marseille, LMA, 4, Impasse Nikola Tesla, CS40006, Marseille 13013, France, cristini@lma.cnrs-mrs.fr), Thierry Garlan (Service Hydrographique et Océanographique de la Marine (SHOM), Brest, France), Xavier Demoulin (Méthodes Acoustiques de REconnaissance de l'Environnement (MAREE), Ploemeur, France), Olivier Morio (Service Hydrographique et Océanographique de la Marine (SHOM), Brest, France), Anne Deschamps, David Ambrois (Université Côte D'Azur CNRS, IRD, OCA, Géoazur, Valbonne, France), and Eric Beucler (Laboratoire de Planétologie et Géodynamique (LPG) CNRS, UMR 6112, Nantes, France)

In order to study the impact of the potential explosion of WWII unexploded ordnances which are commonly found along the French Mediterranean coast, a series of underwater explosions were conducted in December 2018 in the Rade of Hyeres in France. These explosions were realized within the framework of the POSA project, led by SHOM, and which includes LMA, Géoazur and LPG Nantes which addresses the upstream hazard management issue of such counter-mining operations in the marine field. From topographical and sedimentary measurements performed in this area, physical and geometrical characteristics of the marine seabed have been carefully selected to serve as input data for numerical simulations of seismo-acoustic wave propagation from the source to several seismometers deployed on the coast of Hyeres and surrounding islands. Numerical simulations were conducted using a spectral-element method in the time domain. The impact of the explosive device (source) charge and its location (on the seabed or in the water column), together with the impact of the marine environment properties, on the simulated signals have been studied. The source time function is estimated from *in situ* acoustic signals recorded by hydrophones located around the detonation location. The seismic data recorded on the coast are used to calibrate the numerical simulation and to evaluate the impact of different marine environments with special attention to the influence of the characteristics of the sediment layer.

2:30

2pUW7. Broadband modeling of mid-frequency transmission measurements. Darrell Jackson (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, drj@apl.washington.edu), Dajun Tang, Eric I. Thorsos, Brian T. Hefner (Appl. Phys. Lab, Univ of Washington, Seattle, WA), Michael B. Porter, and Laurel Henderson (Heat, Light, & Sound Reserch, Inc., La Jolla, CA)

Acoustic transmission measurements are commonly used to infer seafloor properties needed in simulations to predict sonar performance. Successful inversion of transmission data requires use of models for acoustic seafloor interaction as well as propagation. In the present work, a time-domain model for broadband reflection (Thorsos *et al.*, this meeting) is combined with the BELLHOP ray-tracing code. The model is developed to handle explosive sources. Plausible seafloor geoacoustic parameters are explored using this model. Issues of uniqueness and inadequacies of the model will be discussed. [Work sponsored by ONR.]

2:45

2pUW8. Theoretical analysis of bottom reflected broadband waveforms. Eric I. Thorsos (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, eit@apl.washington.edu), Darrell Jackson, Dajun Tang, Brian T. Hefner (Appl. Phys. Lab, Univ of Washington, Seattle, WA), Michael B. Porter, and Laurel Henderson (Heat, Light, & Sound Reserch, Inc., La Jolla, CA)

Bottom reflection of broadband waveforms from explosive sources leads to pulse distortion, including the presence of a precursor to the main pulse, a topic that goes back to the 1950s [Arons and Yennie, *JASA* **22**, 231–237 (1950)). A theoretical analysis of broadband bottom reflection can provide valuable insights into observed waveform distortion and its dependence on the grazing angle and sediment properties. When the attenuation in the bottom can be approximated as varying linearly with frequency, the magnitude of the reflection coefficient is independent of frequency, but with a common phase shift over positive frequencies (and with the opposite phase shift over negative frequencies). Combining these contributions leads to the pulse distortion . Variations of the sediment properties that cause the same phase shift at a given grazing angle produce the same pulse distortion. Conversely, inferring a phase shift from the pulse distortion can be used to place constraints on the inversion of sediment properties. Examples will be discussed. [Work supported by ONR.]

3:00–3:15 Break

3:15

2pUW9. Investigation on cutoff frequency and radial dependence of the sound field inelastic fluid-filled pipelines. Dajing Shang (Underwater Acoust. Eng. Dept., Harbin Eng. Univ., No.145, Nantong St., Nangang Dist., Harbin City, Heilongjiang Province 150001, China, shangdajing@hrbeu.edu.cn), Qi Li, and Jiapeng Song (Underwater Acoust. Eng. Dept., Harbin Eng. Univ., Harbin, China)

Elastic fluid-filled pipeline system has been widely used in both life and military. It is of great significance for vibration and noise reduction of underwater vehicles to investigate the acoustic propagation characteristics of elastic fluid-filled pipeline system. The propagation path of low frequency noise in the pipeline is related to the radial dependence of the sound field in the pipeline. In this paper, the frequencies of normal waves in an elastic fluid-filled pipeline were calculated, and the axial and radial dependences of sound fields were analyzed. An experimental system for investigating acoustic propagation in a fluid-filled PE pipeline was constructed and verified the theoretical results. The cut-off frequencies of the normal waves and the radial dependence of the sound in the pipeline were carefully investigated. The results show that there is a sound cut-off phenomenon in the elastic fluid-filled pipeline; the radial dependence of the sound field in the pipeline conforms to the Bessel function law.

3:30

2pUW10. Modelling sound propagation in the ocean from multiple sound sources. Ray Kirby (Ctr. for Audio, Acoust. and Vib., Univ. of Technol., Sydney, Broadway, Ultimo, New South Wales 2007, Australia, ray.kirby@uts.edu.au)

Modelling sound propagation in ocean waveguides normally assumes that a sound source can be represented by a single, idealised, point source. However, the ocean environment contains many different and complex sound sources and it may not always be appropriate to represent these as a point source, especially in the acoustic near field and/or shallow water environments. Accordingly, sound propagation in an ocean waveguide is examined here using a numerical model to accommodate multiple point and dipole sources that are arranged vertically in the ocean. Predictions are obtained using a semi analytic finite element approach, which is a normal mode model that computes eigenmodes propagating in the range direction. This enables the addition of multiple point sources, as well as depth dependent fluid properties, and compressional and shear waves in the seabed. An investigation is then undertaken to ascertain where the acoustic near field ends for complex sources in order to examine the appropriateness of the assumption of an idealised point source for these more complex problems.

3:45

2pUW11. Coupling strategies for the simulation of noise level distribution in the ocean. Paul Cristini (Aix Marseille Univ., CNRS, Centrale Marseille, LMA, 4, Impasse Nikola Tesla, CS40006, Marseille 13013, France, cristini@lma.cnrs-mrs.fr), Roberto Sabatini (Physical Sci., Embry-Riddle Aeronautical Univ., Daytona Beach, FL), Claire Noel, Simon Marchetti, and Jean-Marc Temmos (semantic-ts, Sanary, France)

Many different types of sources of noise are now present in the ocean which all contribute to the increase of noise level and have to be addressed. As a consequence, the accurate numerical simulation of acoustic wave propagation in the ocean is very important in order to be able to predict the distribution of noise level generated by all types of sources which will help to assess their impact on marine life. Many acoustic propagation models are available. Under some assumptions, they all are able to propagate the sound field over long distances. Nevertheless, the underlying hypothesis behind these models often leads to some lack of accuracy. This is the price to pay to be able to propagate over long distances with a reasonable computational cost. On the other hand, accurate numerical modeling requires a computational burden which can be heavy and not always realistic. We present a tool which aims at being a compromise between accuracy and feasibility for the generation of acoustics maps of noise distribution in the ocean where we choose to evaluate the sound field in the vicinity of the source by means of an accurate numerical method such as the finite spectral-element method and to couple the results provided by this numerical method to more classical numerical models like ray tracing or the parabolic equation method.

4:00

2pUW12. Variational formulation for sensitivity analysis of a two-dimensional energy flux model. Mark Langhirt (ARL Penn State, Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mal83@psu.edu), Charles W. Holland, and Sheri Martinelli (ARL Penn State, State College, PA)

Energy flux propagation models have demonstrated promising results in efficiently approximating both incoherent and coherent field intensities in range dependent environments [Holland, *J. Acoust. Soc. Am.* **128**, 2596 (2010); Harrison *J. Acoust. Soc. Am.* **133**, 3777 (2013)]. The efficiency of energy flux models is advantageous when applied to solving complex problems in environments that would typically require much more computational effort with other propagation models. Some environmental range dependencies that have been investigated with the energy flux model include sound velocity, bathymetry, and sediment properties. Though comparisons have been made of static solutions between energy flux and other propagation models, there has been limited investigation of energy flux models in regards to variational sensitivity to perturbations in the environment. The applications for variational analyses are broad in a general sense, but the core goal is to capture the robustness of the forward model to perturbative inputs. This study seeks to establish the ground work for defining an adjoint-model-based sensitivity analysis of a 2-D range-dependent energy flux model.

4:15

2pUW13. Comparing analytical and machine-learning techniques for predicting acoustic fields in ideal single- and multi-path environments. Brandon M. Lee (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, leebm@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Neural-network-based machine learning provides a modern solution paradigm for a wide variety of problems. Trained neural networks can provide answers quickly because the primary computational effort is front-loaded into network training. This presentation provides the results of an investigation into the neural-network-development effort necessary to replace point-source Helmholtz-equation solutions in axisymmetric single-path, two-path, and multi-path environments having constant sound speed. Here, the acoustic and environmental input parameters are provided to the neural network as they would be to a propagation simulation. The single-path (free space) environment involves four parameters (sound speed, frequency, receiver range, and receiver depth). The two-path (Lloyd's mirror) environment adds a reflecting surface and a fifth parameter, source depth. The multipath (ideal waveguide) environment further adds a second (deeper) reflecting surface and a sixth parameter, waveguide depth. In all cases, neural network training data and performance comparisons are developed from well-known analytical Helmholtz-equation solutions. Uniform and non-uniform sampling strategies for neural-netwok training are considered for frequencies in the 100s of Hz, depths up to 200 m, and ranges up to 2 km for sound speeds near 1500 m/s. Comparisons emphasize acoustic-field amplitude. Extensions of these results to acoustic-field phase and more than three propagation paths are considered.

4:30

2pUW14. Remote acoustic illumination using time reversal and a surface ship. Hee-Chun Song (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 8820 Shellback Way, Spiess Hall, Rm. 448, La Jolla, CA 92093-0238, hcsong@ucsd.edu), Gihoon Byun (Scripps Inst. of Oceanogr., Univ. of California, San Diego, San Diego, CA), and Jea Soo Kim (Korea Maritime and Ocean Univ., Busan, South Korea)

Time-reversal (TR) transmission of the Green's function between a time-reversal mirror (TRM) and a probe source (PS) in an acoustic waveguide produces a spatio-temporal focus at the PS location. The TR focus then behaves as a virtual point source in the outbound direction with respect to the TRM. Further, a collection of adjacent TR focuses may constitute a virtual source array (VSA) that can serve as a remote platform, redirecting the focused field to a selected location beyond the VSA for which the Green's function is not available *a priori*. The practical limitation to the VSA implementation, however, is the requirement of a PS at multiple adjacent locations to obtain the Green's functions between TRM and VSA. Alternatively, this work proposes to utilize a surface ship radiating broadband noise as a PS in conjunction with the waveguide invariant theory, instantly generating a horizontal VSA. The feasibility of remote acoustic illumination using a ship and a TRM is demonstrated using numerical simulations in shallow water.

4:45

2pUW15. High fidelity modeling of pile driving acoustic energy. Kevin D. Heaney (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com), Kerri Seger (Appl. Ocean Sci., Santa Monica, CA), Michele B. Halvorsen (CSA, Stuart, FL), Michael A. Ainslie (JASCO, Eschborn, Germany), Marten Nijhof, and Roel Muller (TNO, Den Hague, The Netherlands)

Pile driving for the installation of wind farms has become a recent topic of interest as we seek to learn how to protect the marine environment while energy efficient offshore turbines are installed. For environmental compliance, propagation models for spreading from a point source are sometimes used to evaluate impact volumes. Use of a point source is problematic in that a pile (either hammered or shaken using a vibratory source) fully extends through the water column and cannot be considered a point source.

In this paper we present a hybrid model solution to evaluate the sound field at close and far ranges. First, a finite element model, consisting of the hammer, pile, the nearby seawater and the sediment, is used to computed the acoustic pressure field at a short range ($\sim$3 m). The sound pressure field versus depth and frequency at this range is used as the starter field for the Parabolic Equation and the field is then propagated to range, including range dependent environments. Impact volumes, for various marine species hearing groups are computed. [Work supported by BOEM.]

5:00

2pUW16. Application of a coupled elasto-acoustic-damage system to acoustic emissions from sea ice fracture. Jonathan Pitt (Virginia Tech, 900 N. Glebe Rd., Arlington, VA 22203, jonathanspitt@gmail.com)

Sea ice fracture and the resultant acoustic emissions have been studied extensively through observation, experiments, and theoretical analysis. This work demonstrates the applicability of a novel time-domain method for simulating dynamic damage evolution in a coupled structural-acoustic system for the prediction of acoustic emissions from sea ice fracture. The primary aim of this study is to employ first principles modeling of acoustic emissions from failure, as derived via the theory of continuum damage mechanics, including the transition of the acoustic waves from solid to fluid domains. The overall solution method is designed to be compatible with modern high-performance computing environments. The algorithm is staggered, where first the solution for the dynamic fracture evolution is found with an explicit step, and then the coupled computation of the structural-acoustic system response is computed using the new values of the damage and modified geometry. Code and solution verification of the fully coupled solution algorithm are presented, as are comparison to laboratory experiments and field observations for sea-ice fracture.

5:15

2pUW17. Study on the characteristics of the water-entry sound of low-speed metal sphere. Yihan Yang (Harbin Eng. Univ., No.145, Nantong St., Harbin 150001, China, 201112232@hrbeu.edu.cn), Qi Li, Dajing Shang, and Rui Tang (Harbin Eng. Univ., Harbin, Heilongjiang, China)

In this paper, the underwater acoustic characteristics of low-speed metal spheres are measured by direct measurement from the angle of sound pressure of Initial impact sound and bubble pulsating sound, which mainly includes the research on the impact of ball size and ball entry velocity on water entry sound and the characteristics of bubble radius. Experimental data were obtained through a large number of experiments, and the data were simply processed and then fitted by MATLAB. The accuracy of experimental measurement results is analyzed from the perspective of statistics. The results show that the relationship between sound pressure and particle size, 2/3 power of water-entry velocity are exponential function with a base of 10; the relationship between sound pressure level and particle size, 2/3 power of water-entry velocity are linear function; the relationship between bubble radius and particle size, 2/3 power of water-entry velocity are linear function. The variation of each physical quantities with particle size and water entry velocity obtained in this study can be used for simple prediction. With the further research of the subject, the research results will be further optimized.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics (4:45 p.m.) and Computational Acoustics (4:30 p.m.).

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 are as follows:

Engineering Acoustics (4:45 p.m.)	Stuart
Acoustical Oceanography	Empress
Animal Bioacoustics	Edison
Architectural Acoustics	Windsor
Physical Acoustics	Wilder
Psychological and Physiological Acoustics	Garden
Signal Processing in Acoustics	Empress
Structural Acoustics and Vibration	Spreckels

Committees meeting on Wednesday are as follows:

Biomedical Acoustics	Hanover
Signal Processing in Acoustics	Empress

Committees meeting on Thursday are as follows:

Computational Acoustics (4:30 p.m.)	Stuart
Musical Acoustics	Coronet
Noise	Crystal/Continental
Speech Communication	Regent
Underwater Acoustics	Viceroy

Session 3aAA

Architectural Acoustics: Assembly Space Renovation Challenges I

Joseph A. Keefe, Cochair
Ostergaard Acoustical Associates, 1460 US Highway 9 North, Ste. 209, Woodbridge, New Jersey 07052

David Manley, Cochair
DLR Group, 6457 Frances St., Omaha, Nebraska 68106

Chair's Introduction—8:00

Invited Papers

8:05

3aAA1. Ten acoustically challenged venue renovations, nine solutions. David A. Conant (MCH, Inc., 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, dconant@MCHinc.com)

This paper provides a travelogue through ten performing arts venues in the West that are architecturally or historically important. While nine found happiness, the tenth still struggles mightily with an inconvenient conundrum—an intractable juxtaposition of its acoustical challenges spread across its history, its geographic setting and its name. Where salient, measured data are presented for each along with highlights of the re-design challenges and their specific solutions, several of which were quite novel.

8:25

3aAA2. Case study of the renovation of Sacramento City Community College auditorium, two theaters and twenty-nine classrooms. Steve Pettyjohn (The Acoust. & Vib. Group, Inc., 5765 9th Ave., Sacramento, CA, spettyjohn@acousticsandvibration.com)

A renovation was undertaken of the main auditorium, a small theater, a black box theater and twenty-nine classrooms to bring them into the digital age. These facilities had been designed in the late 1960s with only minor changes to update, for example, the black box theater. The project involved the architectural and mechanical acoustical design of these spaces and the design of the audio-video system for the performance areas and for the classrooms. The focus of this paper is on the renovation of the auditorium. A major change was removing the balcony seating, reducing the seating from 1100 to 600. The space was to be used for programs ranging from plays to rock music to symphonies. This meant that the reverberation time had be variable from less than 1-s to over 2-s. Panels were designed to swing on the side walls into two positions to achieve the variable reverberation times. The sound system for the auditorium, specifically the mixing board, had to be designed around that used for classroom instruction. Financial problems created large problems as did a decision to remove the specifications for vibration isolation of the HVAC system. A summary of results will be presented.

8:45

3aAA3. Of body shops, shuls, and shiraz. Jesse J. Ehnert (Arpeggio, 1060 Mercer St., SE, Atlanta, GA 30316, jehnert@arpeggiollc.com)

Completed in March of 2013, the new worship facility for The Congregation or Hadash in Atlanta, Georgia is an adaptive reuse of a vacant automotive paint and body repair facility constructed primarily of concrete, masonry, and metal. The synagogue, which was designed to accommodate 400 families, contains a sanctuary, social hall, chapel, offices, and classrooms along with exterior courtyards and gardens. The focal point for the synagogue is the 450-seat sanctuary which, due to the multinational nature of the congregation, had to promote a high degree of speech intelligibility. Additionally, live music plays a prominent role in typical services; therefore, the acoustics of the room needed to achieve a compromise between the often conflicting demands of these two different types of program. In addition to the architectural realities presented by the existing industrial facility, the building's location with a high school football stadium located approximately 200' away provided a significant challenge in the form of environmental noise, particularly on Friday nights when games (with associated marching bands and public address system usage) and services were likely to be concurrently held. However, perhaps the greatest challenge posed was due to client expectations, based on what they had been accustomed to. Sometimes venues need to age like good wine.

9:05

3aAA4. A case study of a 600-Seat Community College Theater renovation. Joseph A. Keefe (Ostergaard Acoust. Assoc., 200 Executive Dr., Ste. 350, West Orange, NJ 07052, jkeefe@acousticalconsultant.com)

The Theater at Monroe Community College in Rochester, NY, was a space with low reverberation time, undesirable focused sound reflections, excessive HVAC system noise, and a haphazard collection of sound system equipment. Design of a renovation commenced in 2015, with strong focus on acoustical improvements. Visually stunning wood "ribs" create aimed reflecting surfaces, promote diffusion, and increase room volume. The ribs converted a previously unbroken ceiling layer to a layer transparent to sound transmission; a gypsum board "lid" was specified to attenuate rainfall noise while maintaining as much room volume as practical. Traditional application of sound absorption to avoid echoes was not desirable in this space; acoustically transparent railings and out-of-plumb rear walls were designed to meet architectural needs while affecting reverberation time minimally. New HVAC ductwork was specified throughout the space, but existing ductwork remained in the mechanical room and in the basement below the theater. The renovation was completed in 2017. The mid-frequency reverberation time increased from 0.7 to 1.1 seconds, while the HVAC system sound level was reduced from a maximum of NC-40 to NC-30. A new sound system was designed and specified that results in uniform high-quality coverage for all seating areas, including the control balcony.

9:25

3aAA5. Town School for boys, theater/gymnasium. Jason R. Duty (Charles Salter Assoc., 130 Sutter St., Fl. 5, San Francisco, CA 94104, jason.duty@cmsalter.com)

During their expansion project, the Town School for Boys in San Francisco created an underground concrete shell for a future gymnasium/450 seat multi-purpose theater with a rooftop playfield above it. They initially planned for a junior high school-sized basketball court at 74 feet $\times$ 42 feet. Years later, the project build-out began and design challenges were encountered, including: (1) The school requested a high school-sized basketball court (84 feet $\times$ 50 feet). (2) Music classrooms above the stage area were added to the project scope. (3) The project was immediately adjacent to residences. (4) Large ventilation openings on the concrete roof meant that HVAC system noise at the property lines and noise intrusion from the playfield were both issues. (5) Impact noise from the playfield was audible in the shell space. (6) The school wanted to concurrently use the playfield and the theater. (7) A late gift from a donor at the beginning of construction triggered a major redesign of the room acoustics. This paper will cover the history of the project, as well as the challenges and design solutions, which included stepped concrete floating floors, spring isolated ceilings, complicated mechanical system design, and variable acoustics.

9:45–10:00 Break

10:00

3aAA6. Metropolitan metamorphosis: A tale of two theater's journeys to acoustical acceptance. Robert M. Tanen (Acoust., Metropolitan Acoust., 1628 John F. Kennedy Blvd., Ste. 1902, Philadelphia, PA 19103-2116, r.tanen@metro-acoustics.com) and Scott Hulteen (Acoust., Metropolitan Acoust., Philadelphia, PA)

Not every assembly space (be it a lecture hall, theater, multi-purpose room, GymaNataCafaConfraTorium, etc.) has the prestige or budget of Concertgebouw in Amsterdam, or the Kimmel Center, here in Philadelphia. Many of the day-to-day jobs that are encountered in the consulting field are renovations of existing spaces at local educational facilities or businesses, and they require the same love and attention but have different restrictions and challenges. Some roadblocks and hurdles include tight budgets, accelerated schedules, space constraints, and/or complex (sometimes fragile) bureaucratic procedures. Today we will look at two recent renovations: one a transformation of a space-restricted, dark and sterile school performance space into a modern, clean and functioning multi-purpose theater; the other a poorly laid out, non-functioning, outdated auditorium (better suited for a fall-out shelter) into a warm, comfortable, accommodating and inspiring space. These two case-studies outline the challenges encountered, solutions developed and heartwarming transformation details that made the renovations a success.

10:20

3aAA7. Renovation of a middle school older gymnasium to provide a multipurpose performance and assembly space. Robert C. Coffeen (RC Coffeen Consultant in Acoust. LLC, 4721 Balmoral Dr., Lawrence, KS 66047, bob@rccoffeen.net)

St. John School in Lawrence, Kansas found it possible and necessary to construct a new gymnasium of proper size on the north side of their school building. The construction of this new gymnasium was completed in early 2018. The new gymnasium is partially adjacent to an older and smaller gymnasium that has a small stage in one end. This older gymnasium was renovated to address aesthetic, acoustic, noise, and theatrical issues so as to provide a multipurpose performance and assembly space. The renovated gymnasium was opened in October 2018 and it has proved to be a successful multipurpose performance and assembly facility.

10:40

3aAA8. A collection of insights gained through assembly space renovation projects. Joseph F. Bridger (Stewart Acoust. Consultants, 7330 Chapel Hill Rd.,Ste. 101, Raleigh, NC 27607, joe@sacnc.com), Siddharth Mahajan, John Gagliardi, Fred Schafer, and Noral D. Stewart (Stewart Acoust. Consultants, Raleigh, NC)

Renovations of assembly spaces are carried out due to a large variety of reasons. These could range from a desire for improved aesthetics, use of up-to-date technologies, recovery from damages or even repurposing for a different purpose. As the project does not start off with a clean slate, it brings many constraints that may prevent acousticians from using their conventional methods of solving certain problems and make them return to the first principles of such problem-solving. The authors believe that some of the most interesting and educational projects they have been involved in were such renovation projects. In this paper, they present a collection of valuable insights gained by their consulting firm in the recent past while working on assembly space renovation projects.

11:00

3aAA9. Willson Auditorium renovation. Sean Connolly (Big Sky Acoust., PO Box 27, Helena, MT 59624, sean@bigskyacoustics.com)

As a small-city community theater, the 1100 seat Willson Auditorium is a multi-use facility in every sense of the word. This former high school auditorium, where Gary Cooper performed as a student, is now home to performances by the local symphony, an opera company, ballet companies, traveling amplified acts, and high-school and middle-school music and drama students. This case study discusses how the distribution of sound-absorptive material and the shapes of surfaces used in the auditorium provided less than ideal acoustical characteristics and weak projection of sound from the stage for many of the users. The measures taken to correct those issues as part of a renovation will also be discussed, including an increase in volume and redistribution of sound-absorbing and sound-reflecting surfaces, that prompted the school district music director to say the teachers "are going to freak out when they hear the way it sounds."

11:20

3aAA10. Victoria Hall Theatre—Almost ready for prime time. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

The Victoria Hall Theatre was built in 1921 as a lively church in the center of Santa Barbara, California, but matured into a tired theatre in the center of the downtown Arts District. Initial considerations whether to renovate began with a study about minimizing intrusion of loud dance music from a neighboring open-deck club. Thereafter, the renovation took off, with major changes including an entirely new stage house, and replacing a problematic balcony with a continuous floor plan with great sightlines for its 294 seats. Exterior noise isolation is excellent considering its wood construction, and interior finishes promote great clarity and improved intimacy. However, the decision to reuse the six old air handling units in the attic, compounded by some questionable mechanical engineering omissions, undermined the theatre's potential to make sound system reinforcement essentially unnecessary. This paper will review various renovation decisions, and the several steps to quiet the renovated HVAC to its current moderate levels, plus the follow-up that almost provided authorization to further quiet the system.

Session 3aAB

Animal Bioacoustics: Urban Noise: Its Effects on Animals' Acoustic Communication

Benjamin N. Taft, Cochair
Landmark Acoustics LLC, 1301 Cleveland Ave., Racine, Wisconsin 53405

Kurt M. Fristrup, Cochair
Natural Sounds and Night Skies Division, National Park Service, 1201 Oakridge Drive, Suite 100, Fort Collins, Colorado 80525

Chair's Introduction—8:00

Invited Papers

8:05

3aAB1. Immediate signaling flexibility in the model system *Zonotrichia leucophrys nuttalli*. Katherine E. Gentry (George Mason Univ., 4800 University Dr., Fairfax, VA 22030, gentry.ke@gmail.com) and David Luther (George Mason Univ., Fairfax, VA)

Background noise can interfere with acoustic communication and subsequently influence signaling behavior. Immediate signaling flexibility (ISF) is a context-dependent form of behavioral plasticity that allows animals to temporarily change their acoustic behavior in response to noise fluctuations and potentially improve the chances of successful communication in noisy environments. However, the adaptive value of ISF is ultimately contingent on the response of the intended receiver, and there are differential effects on receiver response depending on which signal component is modified. Here, we describe ISF in the model system *Zonotrichia leucophrys nuttalli* using results from a noise broadcast experiment conducted in urban and rural locations. Three types of noise broadcasts were incorporated into the experimental design to test whether plasticity of song behavior, if observed, was dependent on the spectral qualities of the experimental noise. We predicted that all birds would exhibit ISF, but vocal plasticity would be greater in noisier territories. Instead, we found that only urban males displayed ISF by decreasing song bandwidth in response to experimental noise. Song modification did not change with type of noise broadcast or territory background noise, nor did it involve the temporary adjustment of trill structure traits used by receivers to assess vocal performance.

8:25

3aAB2. Anthropogenic noise and social context affect vocal plasticity in a common urban passerine. Erin E. Grabarczyk (Biological Sci., Western Michigan Univ., Kalamazoo, MI), Maarten J. Vonhof (Inst. of the Environment and Sustainability, Western Michigan Univ., Kalamazoo, MI), and Sharon A. Gill (Biological Sci., Western Michigan Univ., 1903 West Michigan Ave., Kalamazoo, MI 49008, sharon.gill@wmich.edu)

Vocal communication shapes animal social networks, connecting individuals over space and time via information, facilitating mate attraction and resource defense. Despite evidence that both the physical and social environment affect signaling behavior, few studies consider variation in individual responses to environmental change within a social context. We test the hypothesis that male House wrens (*Troglodytes aedon*) plastically adjust songs and song sections structured for short-and long-distance transmission in response to change in their immediate noise environment, but that both social context and noise affect singing patterns at the population level. We recorded paired males prior to clutch initiation, quantified ambient noise in the moments before singing, and define social context within pairs as female fertile status and between males as number of conspecific neighbors. Among males, adjustment patterns varied depending on transmission properties, social context, and noise. In response to immediate change in noise, males plastically adjust some, but not all, song traits. We show that not all males adjust signals in the same way, and that consideration of social context and signal function are crucial for understanding variation in signal structure. This is an essential step towards understanding how both the social and physical environment may drive selection on vocalizations.

8:45

3aAB3. Singing in the city: Trade-offs between signal salience and vocal performance in the presence of anthropogenic noise. Jennifer N. Phillips (Biological Sci., California Polytechnic State Univ., 1 Grand Ave., San Luis Obispo, CA 93407, jnphilli@calpoly.edu), David Luther (George Mason Univ., Fairfax, VA), and Elizabeth Derryberry (Univ. of Tennessee, Knoxville, TN)

Many animals use acoustic signals to communicate. Signals specifically used to compete for or select a mate are under sexual selection. Selection therefore favors signaling traits that indicate competitive ability or mate quality. For example, performance of signaling traits that are costly to produce, such as those that are physically limited by the animal's size or coordination are more likely to convey honest information about the sender. Selection also favors signaling traits that maximize the transmission of this information in a given

environment. For example, other sounds in the environment (the soundscape) can reduce the ability of individuals to detect or decode information in the signal. However, individuals may not always be able to maximize both performance and transmission of a signal. This leads to two critical questions—when faced with this trade-off, which aspect of the signal do senders maximize and how does this decision affect communication? I address these questions using a well-studied sexual signal—bird song—in anthropogenic soundscapes, which are evolutionarily recent selective environments. Here, I tested whether vocal performance functions in male-male competition in White-crowned Sparrows (*Zonotrichia leucophrys*), and how assessment of performance varies across different urban and rural soundscapes in San Francisco and Point Reyes, California. I also investigate how soundscapes go beyond individual behavior and affect survival of these populations.

9:05

3aAB4. Do aircraft events alter the vocal behavior of passerine bird communities? Allison Injaian (Lab of Ornithology, Cornell Univ., 159 Sapsucker Woods Rd., Lansing, NY 14850, asi27@cornell.edu)

Anthropogenic noise alters ambient sound levels even in the most protected habitats in the US, with airports being a major contributor of noise. Birds breeding near airports may alter their vocal behavior, perhaps due to a reduction in communication space (i.e., masking). Alterations in vocal behavior may result in a reduced ability to attract a mate or maintain a territory. Here, we use passive acoustic monitoring to assess the effects of aircraft noise on the vocal behavior of common bird species, (e.g., wood thrushes, red-winged blackbirds, etc.) in Ithaca, NY, USA. We measured total time spent singing immediately before and after plane takeoffs for birds breeding at varying proximities to the airport and associated flight paths. Data were collected between 0600 and 0700 h (peak of dawn chorus) from May–July 2017. If the total time spent singing differs based on aircraft status (before/after aircraft event) and noise exposure (peak amplitude associated with aircraft events), our results would suggest that aircraft noise alters dawn chorus. Further, if our data show species-specific impacts of noise, characteristics such as migratory status and/or vocalization frequency (pitch) may be important considerations when predicting noise impacts on birds.

9:25

3aAB5. Exploring the effects of aircraft noise on bioacoustic activity in national parks. Nathan Kleist (Fish, Wildlife, and Conservation Biology, Colorado State Univ., Fort Collins, CO) and Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

In the course of measuring and analyzing environmental acoustic conditions in U. S. National Park units, a large number of acoustical recordings have been aurally reviewed by trained technicians who noted the presence of a wide variety of sound sources. For this presentation, we analyzed changes in bioacoustic activity in relation to sounds from nearby people and distant aircraft. A central question for our analysis was the relevant time scale of exposure, as it is plausible that natural choruses are affected both by the immediate stimuli they experience as well as their recent history of exposure to similar stimuli. The large sample size—600 000 observations—enabled us to investigate data-driven models that document the decay of effects with increasing time lag between exposure and response.

9:45–10:00 Break

Contributed Papers

10:00

3aAB6. A comparison of humpback whale singing and ambient noise cycles in the Gulf of Tribugá, Chocó, Colombia. Laura Valentina-Huertas, Maria Paula-Rey (Universidad Pontificia Javeriana, Bogota, Colombia), Kerri Seger (Appl. Ocean Sci. LLC, 2127 1/2 Stewart St., Santa Monica, CA 90404, kerri.seger.d@gmail.com), Christina E. Perazio (Macuaticos, Biddeford, ME), Natalia Botero (Universidad Pontificia Javeriana, Medellin, Colombia), Valeria Gonzalez (Pacifico Travel, Buenos Aires, Argentina), and Andrea Luna-Acosta (Universidad Pontificia Javeriana, Bogota, Colombia)

To monitor potential changes in the humpback whale (*Megaptera novaeangliae)* singing cycle, typical singing intensities, and acoustic behaviors requires a baseline understanding of singing activity in the context of an acoustic environment that is minimally disturbed. The Gulf of Tribugá in the Colombian Pacific (the breeding ground for Stock G) was named a Hope Spot in June 2019 as a local conservation campaign against the Colombian government's plans to build an international port here. To document this minimally disturbed acoustic ecosystem, data were recorded at Morro Mico from October to November 2018. An Ecological Acoustic Recorder (EAR) collected data on a 33.3% duty cycle at a 15 kHz sampling rate. Power spectral densities were explored across several bands up to 6250 Hz and were used to rank sound source influence and to detect diel and lunar cycles. Spectrograms in Raven Pro 1.5 were used to manually classify sounds into six groups: dolphins, humpback whales, fish, wind and rain, snapping shrimp, and boats. These sound source time series combined with power spectral density and humpback whale song intensity cycles now serves as the first year of a baseline study in preparation to document the effects of a pending major construction project.

10:15

3aAB7. Killer whale (*Orcinus orca*) hearing in noise. Brian K. Branstetter (National Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106, brian.branstetter@nmmpfoundation.org), Kristi Meze-Burtis, Mike Felice (Sea World San Diego, San Diego, CA), James J. Finneran (US Navy Marine Mammal Program, San Diego, CA), and Todd Robeck (Sea World San Diego, San Diego, CA)

Killer whales (*Orcinus orca*) are an apex predator, the largest delphinid odontocete, and have the widest cosmopolitan distribution of all cetacean species. These long-lived animals can be found in life-long matrilineal groups, where fitness-related activities such as cooperative hunting rely on sophisticated group coordination, often mediated by acoustic communication. The effects of anthropogenic noise on the fitness of killer whales is a growing concern, especially for the endangered population of southern resident killer whales. Although modest progress has been made in determining the killer whales' behavior in response to noise (e.g., Lombard effect), almost nothing is known about how these animals hear in noisy environments. Here, what is known about killer whale hearing is summarized, including behavioral and electrophysiological audiograms, and resulting auditory weighting functions. In addition, new critical ratio data are presented and compared with other odontocete species. These preliminary data are evaluated in a communication space model to predict negative impacts of anthropogenic noise.

10:30

3aAB8. How acoustics informs understanding of foraging behavior and effects of vessels and noise on killer whales. Marla M. Holt (Conservation Biology Div., NOAA NMFS Northwest Fisheries Sci. Ctr., 2725 Montlake Blvd East, Seattle, WA 98112, Marla.Holt@noaa.gov), Jennifer B. Tennessen (Under Contract with Lynker Technologies, NOAA Northwest Fisheries Sci. Ctr., University Park, PA), Brad Hanson, Candice Emmons (Conservation Biology Div., NOAA NMFS Northwest Fisheries Sci. Ctr., Seattle, WA), Deborah Giles (Friday Harbor Labs., Univ. of Washington, Friday Harbor, WA), Jeffery Hogan (Cascadia Res. Collective, Olympia, WA), Brianna M. Wright (Pacific Biological Station, Fisheries and Oceans Canada, Nanaimo, BC, Canada), and Sheila Thornton (Aquatic Ecosystem & Marine Mammals Section, Fisheries and Oceans Canada, Vancouver, BC, Canada)

Foraging in toothed whales and dolphins is fundamentally tied to the use of sound. Resident-type killer whales (*Orcinus orca)* use echolocation to locate and capture fast-moving salmon and other fish prey. In addition to prey availability, disturbance from vessels and noise is a threat to the endangered Southern Resident killer whale population given considerable levels of commercial shipping, fishing, whale-watching and recreational vessel traffic in urban waterways that the whales use for feeding. In this study, we utilized suction cup-attached digital acoustic recording tags (DTAGs) to (1) describe whale acoustic and movement behavior during different phases of foraging that can be differentiated from other behaviors, (2) investigate vessel and noise effects on behavior and foraging outcomes in the endangered population, (3) compare foraging behavior between the endangered population that is struggling with population recovery and another population (Northern Resident killer whales) that is increasing in numbers, and (4) characterize diel patterns of foraging and other behaviors to describe their full activity budget and inform management of vessel traffic and noise during urban expansion along the Pacific Northwest coast of North America. This presentation will highlight results to date and implications for the conservation and management of marine protected species.

Invited Paper

10:45

3aAB9. Anthropogenic noise and the many acoustic modalities of terrestrial invertebrates. Maggie Raboin (Environ. Sci., Policy and Management, Univ. of California Berkeley, 130 Mulford Hall, Berkeley, CA 94720, maggie.raboin@berkeley.edu) and Damian O. Elias (Environ. Sci., Policy and Management, Univ. of California Berkeley, Berkeley, CA)

Anthropogenic noise is widely recognized as an issue of environmental concern. However, research to date has focused on the impacts of far-field airborne noise (i.e., pressure waves) on vertebrates while the impact of noise on invertebrates, and the other acoustic modalities they rely on, has received little attention. Although some terrestrial invertebrates use far-field sound for communication, the majority rely on near-field airborne and substrate-borne sound (i.e., particle motion and vibrations, respectively). We discuss what little information is known about the impact of anthropogenic noise on invertebrates and make predictions based on studies of invertebrate bioacoustics. Furthermore, we discuss the ways that noise in each acoustic modality (far-field airborne, near-field airborne, and substrate-borne) might affect communication of terrestrial invertebrates and highlight the most pressing lines of inquiry for future research.

Contributed Papers

11:05

3aAB10. Effects of rail vibrations on humans and mitigation solutions. Alexander C. Born (Getzner USA, Inc., 8720 Red Oak Blvd, Ste. 400, Charlotte, NC 28217, alexander.born@getzner.com)

The impact of rail vibrations on humans is becoming a more prominent issue as buildings are being built closer and closer to rail lines due to limited development space. I will cover this impact along with solutions to prevent it.

11:20

3aAB11. Influence of resonance-driven bubble clouds on fine-scale behaviour of common carp (*Cyprinus carpio*). Nicholas Flores Martin (Faculty of Eng. and Physical Sci., Univ. of Southampton, Water and Environ. Eng. Group, University of Southampton, Southampton, United Kingdom SO16 7QF, United Kingdom, nicholas.floresmartin@soton.ac.uk), Timothy G. Leighton, Paul R. White (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom), and Paul S. Kemp (Faculty of Eng. and Physical Sci., Univ. of Southampton, Southampton, United Kingdom)

Acoustic-bubble barriers show potential as a management tool for fish passage. Although greater efforts are now being made to better understand the fundamentals that underpin reactions of fish to such barriers, the phenomenon of bubble coalescence has not yet been considered. When introducing air underwater smaller bubbles coalesce with successor bubbles at the orifice so that without vibration the bubble which enters the liquid is large. Orifice diameter has historically been used as a measure of bubble size, however as detachment and fragmentation occur unpredictably this method is unreliable, indicating insufficient control over the stimulus generated. This study used lower air flows (6 l min^{-1} m^{-1}) and haptic feedback motors to vibrate the injection nozzles and reduce coalescence. Greater uniformity in bubble size allowed matching of the bubble population to a sound field to which the bubbles were resonant. In a proof-of-concept study, fish response to two resonant acoustic-bubble barriers was tested and compared to two non-resonant acoustic-bubble walls. The scattering and absorption coefficients for all segments of the bubble population were calculated to determine the most important bubble sizes for attenuation. Analysis of fish movement and orientation was used to examine fish behaviour in relation to the sound fields generated.

Session 3aAO

Acoustical Oceanography and Animal Bioacoustics: Bioacoustics and Acoustical Oceanography: 20 Years Later I

Andone C. Lavery, Cochair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 98 Water Street, MS 11, Bigelow 211, Woods Hole, Massachusetts 02543

Kelly J. Benoit-Bird, Cochair
Monterey Bay Aquarium Research Institute, 7700 Sandholdt Road, Moss Landing, California 95039

Invited Papers

8:00

3aAO1. Two decades of progress in active bioacoustics. Timothy K. Stanton (Dept. Appl. Ocean. Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543, tstanton@whoi.edu)

There has been significant progress over the past 20 years in the area of active bioacoustics. Use of multifrequency echosounders has given way to broadband systems. Autonomous vehicles, used as sensor platforms, which were generally at the prototype level 20 years ago are now fully operational and ubiquitous. Methods such as use of echo statistics to infer meaningful biological information have advanced from modeling narrowband signals, simple scatterers, and direct path geometries to broadband signals, randomized elongated scatterers, and long-range sonars. These advances and others will be summarized, as well as looks to the future.

8:20

3aAO2. Mapping the global prey-field: Combining acoustics, optics and net samples to reduce uncertainty in estimates of mesopelagic biomass. Roland Proud (School of Biology, Univ. of St Andrews, Bute Medical Buildings, Queen's Terrace, St. Andrews, Fife KY16 9TS, United Kingdom, rp43@st-andrews.ac.uk), Rudy J. Kloser (CSIRO Oceans and Atmosphere Flagship, Hobart, Tasmania, Australia), Nils Olav Handegard (Inst. of Marine Res., Bergen, Norway), Martin J. Cox, and Andrew S. Brierley (School of Biology, Univ. of St. Andrews, St. Andrews, United Kingdom)

Mesopelagic (200–1000 m depth zone) biomass largely comprises zooplankton, jellyfish, squid and fish. Part of this biomass actively migrates between the surface and the mesopelagic zone on a daily basis, playing an important role in global biogeochemical cycling. When observed using echosounders, aggregated components of this community (largely made up of fish and siphonophores) are known as Deep Scattering Layers (DSLs). DSLs have been used to partition the Global Ocean into biogeographically distinct regions, based on DSL vertical distribution and echo energy, and estimates of mesopelagic biomass have stemmed from this work. These estimates are typically derived using a combination of biological sampling, optics, and active-acoustics, but are often biased (e.g., through trawl-based escapement and avoidance) and uncertain. We have developed a new method that combines multiple strands of observations, both from vessels and ocean probes, to minimise sampling bias and produce accurate estimates of mesopelagic biomass. We apply the method to the specific case of estimating mesopelagic fish biomass in the Tasman Sea, combining observations made from a vessel and a Profiling Lagrangian Acoustic Optical System. We compare our new biomass estimates to previous estimates and ecosystem model predictions. Our new method provides a robust framework for future ocean biomass estimates.

8:40

3aAO3. Biological-physical coupling in a highly advective ecosystem: Through a lens of diel vertical migration. Mei Sato (Inst. for the Oceans and Fisheries, Univ. of Br. Columbia, AERL Bldg., 2202 Main Mall, Vancouver, BC V6T1Z4, Canada, m.sato@oceans.ubc.ca) and Kelly J. Benoit-Bird (Monterey Bay Aquarium Res. Inst., Moss Landing, CA)

A mismatch in sampling resolution between biology and physics has been one of the fundamental challenges in advancing our understanding of ecological processes in the ocean. The recent deployment of cabled observatories enables the collection of active acoustic data with coverage and resolution comparable to physical data, providing an unprecedented ability to observe the behavior of zooplankton that are critical trophic link in the food web. Using multifrequency acoustics, acoustic Doppler current profiler (ADCP), and meteorological sensors deployed by the Ocean Observatories Initiatives (OOI) in the Northern California Current System, we explored small-scale upwelling impacts on diel vertical migration exploiting the frequency and predictability of this ubiquitous behavior. We found that the vertical extent of diel vertical migration changed relative to upwelling intensity. Migration behavior was completely suppressed during strong downwelling periods. Zooplankton migrated throughout the water column during relaxation periods while decreasing their migration distance as the winds shifted to upwelling favorable conditions, potentially increasing their accessibility to food resources. These results suggest that animals adapt their behavior to accommodate changes in the physical environment.

9:00–9:15 Break

Contributed Papers

9:15

3aAO4. Broadband acoustic quantification of mesopelagic zooplankton. Rachel E. Kahn (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 86 Water St., MS 12, Bigelow 109B, Woods Hole, MA 02543, rkahn@mit.edu) and Andone C. Lavery (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

In addition to its critical role in biogeochemical cycling, the mesopelagic zone has the potential to become an important global fisheries resource. A growing number of studies have focused on estimating the global biomass of mesopelagic fishes, but that of mesopelagic zooplankton is far more uncertain. Yet, quantification of zooplankton biomass and distribution is crucial to both understanding the ecology of the mesopelagic zone and informing policy makers on sustainable exploitation. Shipboard, narrowband, volume backscattering measurements—typically performed at 18 and/or 38 kHz—are likely dominated by gas-bearing organisms, hindering accurate assessment of zooplankton biomass. A towed vehicle, Deep-See, was developed and equipped with broadband acoustics from 1–450 kHz, optical sensors, and environmental sensors to address the challenges associated with systematically acquiring holistic, quantitative data to infer mesopelagic biomass. Broadband backscattering spectra (50–160 kHz) collected by Deep-See in Summer 2018 and 2019 off the New England shelf break are used to classify scattering layers into fluid-like zooplankton (e.g., jellies, copepods, krill) or gas-bearing organisms (e.g., fishes, siphonophores), then physics-based scattering models are used to estimate abundance and biomass. Consistent with the acoustic measurements, digital holographic images collected by Deep-See reveal multiple deep scattering layers, some consisting mainly of weakly scattering krill.

9:30

3aAO5. Mesopelagic predator-prey interactions revealed by joint passive and active acoustic observations. Samuel S. Urmy (Monterey Bay Aquarium Res. Inst., 7700 Sandholdt Rd., Moss Landing, CA 95039, urmy@mbari.org), Kelly J. Benoit-Bird, John P. Ryan (Monterey Bay Aquarium Res. Inst., Moss Landing, CA), and John K. Horne (School of Aquatic and Fishery Sci., Univ. of Washington, Seattle, WA)

The ocean's mesopelagic zone is one of the Earth's largest habitats and contains large numbers of animals, playing important roles in aquatic food webs—notably as food for many marine mammal species. While these communities have been observed for decades on echosounders as sound scattering layers (SSLs), their ecological dynamics remain poorly understood. Using a broadband (0-128 kHz) hydrophone and an upward-looking echosounder (38 kHz) at a cabled observatory in Monterey Bay, CA in early 2019, we observed numerous occasions where SSLs abruptly increased their depth during bouts of echolocation clicks from odontocete whales. Clicking bouts occurred 6.1 times per day on average, mostly at night, lasting minutes to several hours. Pacific white-sided dolphins and Risso's dolphins produced most of these clicks. Increases in clicking were significantly cross-correlated with increases in SSL depth. The deepest SSL, centered at 400–500 m depth, dove 15 m during an average clicking bout, with some excursions up to 100 m. Video surveys from a remotely operated vehicle identified juvenile Pacific hake, myctophids, and sergestid shrimp as likely constituents of these SSLs. Our results suggest these animals actively dive to avoid odontocetes, and that fear of predation can restructure mesopelagic ecosystems on short time scales.

9:45

3aAO6. Quantitative examination of data processing effects on species classification from broadband echosounders. Kelly J. Benoit-Bird (Monterey Bay Aquarium Res. Inst., 7700 Sandholdt Rd., Moss Landing, CA 95039, kbb@mbari.org) and Chad M. Waluk (Monterey Bay Aquarium Res. Inst., Moss Landing, CA)

Measurements from broadband echosounders hold much promise for improving the ability to discriminate biological targets in the ocean. However, it remains unclear how effectively the increased bandwidth can be leveraged *in situ* and few clear protocols exist to guide the user in making choices in processing these voluminous data. We used a 45–170 kHz wideband echosounder and a high definition video system to simultaneously observe aggregations of fish and invertebrates *in situ* using a remotely operated vehicle (ROV). The dataset contains a large number of echoes collected in a minimally disturbed context, avoiding any effects of compression/decompression of air-containing target and the identity of the targets was verified at a scale and resolution comparable to the acoustic measurements. Using these data, we quantitatively explore the effects of (1) the chosen frequency resolution, (2) the method and scale of data averaging, (3) available bandwidth, (4) data normalization choices, and the (5) use of volume scattering versus resolved single targets on the ability to discriminate echoes from three species. There was sufficient information in broadband data to accurately discriminate between echoes from known species but the accuracy of classification was strongly affected by the data processing choices.

10:00

3aAO7. How much more informative are broadband compared to narrowband echoes for biological interpretation? Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wjlee@apl.washington.edu), Dezhang Chu (NOAA Northwest Fisheries Sci. Ctr., Seattle, WA), and Stan E. Dosso (School of Earth & Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

The recent availability of commercial broadband echosounders has elicited wide interests in their potentials for enhancing the effectiveness, efficiency, and accuracy of acoustic sensing capability for monitoring mid-trophic level marine organisms. However, despite the significantly improved temporal and spatial resolutions, it remains unclear how the additional spectral information provided by broadband echosounders contribute to achieving these goals. In this study, we use a Bayesian inversion framework to compare the estimation uncertainty between broadband and narrowband echo data for biological model parameters, such as organism length, tilt angle, numerical density and aggregation composition. We employ the Markov Chain Monte-Carlo (MCMC) sampling technique to construct the posterior probability density (PPD) of biological parameters given simulated zooplankton and fish echo data in the form of calibrated volume backscattering strength (Sv). The data are simulated for frequency ranges commonly employed in marine ecological and fisheries surveys. We investigate the changes in PPD in response to variations in echo spectral information, with specific emphasis on the correlation structure among model parameters and whether and how broadband information reduces the uncertainty in inferring biological information from acoustic quantities available from field surveys. [Work supported by NMFS Office of Science and Technology Advanced Sampling Technology Working Group.]

10:15

3aAO8. Two decades of progress in physics-based echo statistics. Timothy K. Stanton (Dept. Appl. Ocean. Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA 02543, tstanton@whoi.edu), Wu-Jung Lee (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), and Kyungmin Baik (Ctr. for Medical Convergence Metrology, Korean Res. Inst. of Standards and Sci., Daejeon, South Korea)

The echo from an aggregation of marine organisms naturally fluctuates from ping to ping as the echosounder is moved over that aggregation due to various interference phenomena and statistical processes. There is important information in the characteristics of the fluctuations, such as numerical density of the scatterers, which can be lost through averaging when calculating volume scattering strength. Physics-based echo statistics involves describing the fluctuations through use of the parameters of the echosounder (signal type, beamwidth), scattering properties of the marine organisms, interference effects of echoes from multiple scatterers, and effects due to the environment such as presence of boundaries and heterogeneities in the medium. Advances over the past 20 years will be reviewed which include describing the statistics for broadband signals, randomized elongated scatterers, and long-range sonars. This presentation will draw from the tutorial recently published by these authors which also contains open access software to calculate the statistics over a wide range of scenarios [*J. Acoust. Soc. Am.* **144**, 3124–3171 (2018)].

WEDNESDAY MORNING, 4 DECEMBER 2019 GARDEN, 7:55 A.M. TO 12:00 NOON

Session 3aBA

Biomedical Acoustics: New Frontiers in Doppler Ultrasound

Alfred C. Yu, Cochair
University of Waterloo, EIT 4125, Waterloo, Ontario N2L 3G1, Canada

Jeffrey A. Ketterling, Cochair
Riverside Research, 156 William St., New York, New York 10038

Invited Papers

7:55

3aBA1. Ultrafast high frequency ultrasound Doppler imaging and its biomedical applications. Chih-Chung Huang (Dept. of Biomedical Eng., National Cheng Kung Univ., No. 1, University Rd., Tainan 70101, Taiwan, cchuang@mail.ncku.edu.tw)

Cerebrovascular disorders are associated with Alzheimer's disease (AD). Preclinical animal study is necessary for understanding AD pathogenesis and determining its optimal diagnosis and treatment strategies. Conventionally, the cerebral vasculature's structure is analyzed through histological staining. However, functional analysis of the cerebral vasculature requires an *in vivo* approach to visualize the blood flow in small animal brains. This paper proposes high-frequency micro-Doppler imaging (HFμDI) technology for mapping mouse cerebral vasculature. Using a 40-MHz transducer enabled *in vivo* visualization of the mouse brain up to 3 mm in depth; furthermore, a minimal vessel diameter of 48 μm could be determined without using microbubbles. Animal experiments determined that the cortical and hippocampal vessel density in young (4-month-old) wild-type mice was similar to that in middle-aged (11-month-old) wild-type mice. However, compared with the vessel density in middle-aged wild-type mice, that in middle-aged mice with AD was significantly lower, particularly in the hippocampus. *In vivo* observation of cerebral vasculature demonstrated the effectiveness of HFμDI for the preclinical study of AD and as a potential tool for AD diagnosis.

8:15

3aBA2. Suppression of *in vivo* flow artifacts in power Doppler through spatial and angular coherence beamforming. Jeremy J. Dahl (Radiology, Stanford Univ., 3155 Porter Dr., Palo Alto, CA 94304, jeremy.dahl@stanford.edu), Byung Chul Yoon, Isabelle Durot, Jessica Ducey-Wysling, Aurélie D'Hondt, Bhavik Patel, Erika Rubesova, Marko Jakovljevic, and You Li (Radiology, Stanford Univ., Stanford, CA)

Power Doppler imaging is traditionally the flow imaging technique utilized clinically in the task of flow detection. Due to the relatively low pulse repetition frequencies used, power Doppler ensemble sizes are small to allow real-time imaging. The small ensemble size, however, makes power Doppler imaging subject to the so-called flash artifact due to limited stationary clutter filtering capabilities. In addition, with the increasing population of overweight and obese individuals, power Doppler imaging is subject to higher amounts of thermal noise and reverberation artifact that pass through the stationary clutter filters. We present coherence beamforming techniques, applied to *in vivo* human imaging, to demonstrate their ability to reduce flow artifacts and noise in power Doppler imaging. Using coherence beamforming techniques in Coherent Flow Power Doppler (CFPD), we show that artifacts due to reverberation and thermal noise are reduced, corresponding to an increase of 7.5 dB in signal-to-noise ratio, in liver imaging of overweight and obese individuals. Similarly, we demonstrate coherence beamforming techniques in CFPD to reduce flash artifacts in videos of the neonatal brain.

8:35

3aBA3. Plane-wave imaging of ocular blood-flow. Ronald H. Silverman (Ophthalmology, Columbia Univ. Irving Medical Ctr., 635 W 165th St., Rm. 711B, New York, NY 10032, rs3072@cumc.columbia.edu), Raksha Urs (Ophthalmology, Columbia Univ. Irving Medical Ctr., New York, NY), Jeffrey A. Ketterling (F.L. Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY), Billy Y. Yiu, and Alfred C. Yu (Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

Blood-flow plays an important role in pathogenesis and progression of many ocular diseases. While optical coherence tomography angiography (OCT-A) has revolutionized depiction of the retinal vasculature, it provides little information regarding flow velocities and cannot visualize the arteries and veins supplying and draining the eye. We implemented plane-wave ultrasound methods to address these shortcomings using the Verasonics Vantage-128 with L22-14 linear array probes. We are performing clinical studies of glaucoma, retinopathy of prematurity, preeclampsia, vascular malformations and tumors. Scans are typically acquired for 3 s, capturing 2-3 cardiac cycles. Compounded data (2–6 angles) acquired at a 1–6 kHz PRF are post-processed using a singular value decomposition filter to suppress stationary structures and power-Doppler images are generated. By selecting areas-of-interest representing specific vessels or the choroid (the vascular tissue underlying and supplying the retina), spectrograms are produced, enabling measurement of flow velocities and resistive indices for each vascular component. We are also performing pre-clinical studies of blood-flow in the rat eye, with and without introduction of contrast microbubbles, and are developing super-resolution methods that approach OCT in resolution for improved depiction of the microvasculature as we develop the rat as a model of glaucoma.

Contributed Papers

8:55

3aBA4. Color Doppler ultrasound twinkling in pathological biomineralizations. Eric Rokni (Penn State Univ., 201 Appl. Sci. Bldg., The Penn State Univ., State College, PA 16802, ezr144@psu.edu) and Julianna C. Simon (Penn State Univ., University Park, PA)

The color Doppler ultrasound twinkling artifact has been observed on some kidney stones as well as various sites in the body where pathological biomineralization can occur. On kidney stones, twinkling was recently attributed to stabilized crevice microbubbles, which suggests that microbubbles may also be present on other minerals. However, it is unknown how crystal structure, composition, and the surrounding environment influences the presence and location of microbubbles and twinkling. Here, crystals found in pathological biomineralization (i.e., uric acid, cholesterol, calcium phosphate, and calcium oxalate), were grown *in vitro*, embedded in a tissue-mimicking phantom, and imaged with a research ultrasound system at 5 and 18.5 MHz. Doppler power was quantified and compared between crystal composition ($n = 5$ per type). We found twinkling was strongest in cholesterol crystals and weakest in calcium phosphate crystals. Micro-computed tomography (μCT) of an *ex vivo* calcium oxalate kidney stone (i.e., a heterogeneous crystal) submerged in water revealed the presence of gas within internal microcracks, which was confirmed by reducing the hydrostatic pressure in a subsequent μCT scan. Future work includes exposing pure *in vitro* crystals to μCT and ultrasound with changes in hydrostatic pressure to evaluate for internal gas pockets that may contribute to twinkling.

9:10

3aBA5. Doppler ultrasound bandwidth imaging: A new technique for mapping unstable flow. Billy Y. Yiu (Elec. and Comput. Eng., Univ. of Waterloo, 250 Laurelwood Dr., Waterloo, ON N2J0E2, Canada, billy.yiu@uwaterloo.ca), Adrian J. Chee (Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada), Guo Tang, Wenbo Luo (Bioprober Corp., Seattle, WA), and Alfred C. Yu (Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

Unstable flow plays an important role in the initiation and progression of atherosclerosis. Yet, mapping flow instability in arteries is challenging because such event is dynamic over both space and time. In this work, we present a new framework called Doppler ultrasound bandwidth imaging (DUBI) that makes use of high-frame-rate plane wave excitation and Doppler bandwidth analysis principles to identify unstable flow regions within an image view. DUBI works by performing autoregressive modeling at every pixel position to estimate the instantaneous Doppler bandwidth, which in principle is broadened by the wider range of velocities attributed to the emergence of unstable flow. DUBI's performance in mapping unstable flow was tested with laminar and turbulent flow conditions in a nozzle-flow phantom. Results showed that DUBI was able to quantify the difference in Doppler bandwidth magnitude (increased from 2.1 to 5.2 kHz) as the flow condition changed from laminar to turbulent. Also, DUBI was found to be able to identify and map the unstable flow region with the image view, outperforming conventional Doppler variance imaging. This observation was substantiated by receiver operating characteristic analysis, in which DUBI achieved a sensitivity of 0.72 and specificity of 0.83 (vs 0.66 and 0.68, respectively, for conventional Doppler variance imaging). We anticipate that DUBI can be applied *in vivo* to obtain useful information for clinical diagnosis of atherosclerosis.

9:25

3aBA6. Detecting kidney stones using twinkling artifacts: Survey of kidney stones with varying composition and size. Benjamin Wood (Radiology, Mayo Clinic, 1523 21st Ave. NE, Rochester, MN 55906, wood.benjamin@mayo.edu) and Matthew W. Urban (Radiology, Mayo Clinic, Rochester, MN)

In recent years, work has been done to understand the mechanisms of Doppler ultrasound twinkling artifacts (TAs) and why they appear over kidney stones. In this work, twinkling artifacts were evaluated as a possible method of locating and characterizing kidney stones. Doppler ultrasound scanning was used to evaluate 47 stones of different types and sizes. An isolated stone study was used to understand the behavior of the TAs while varying transmit voltages. An *ex vivo* kidney study was conducted to determine if TAs were localized to the stones. An *ex vivo* randomized stone placement study was used to evaluate the robustness of detecting and locating stones. The TAs were also shown to be isolated to the stone when placed in an *ex vivo* kidney. The randomized stone placement study showed that this method could find all 47 used stones with only two false positives. A few limitations to this method were found with issues accurately sizing stones as well as difficulties in specificity for characterizing the stones. Further work will be done on these limitations by improving the Doppler acquisition and processing code as well as evaluating the use of TAs in human studies.

9:40

3aBA7. Vector flow imaging using a deep neural network. You L. Li (Radiology, Stanford Univ., 3155 Porter Dr., Palo Alto, CA 94304, leo.you.li@gmail.com), Jessica Ducey-Wysling (Stanford Health Care, Stanford, CA), Aurélie D'Hondt (Radiology, Stanford Univ., Stanford, CA), Dongwoon Hyun (Radiology, Stanford Univ., Palo Alto, CA), Bhavik Patel (Radiology, Stanford Univ., Stanford, CA), and Jeremy J. Dahl (Radiology, Stanford Univ., Palo Alto, CA)

Vector flow imaging (VFI) estimates blood flow velocities in both azimuth and axial dimensions. VFI has promising applications in the characterization of complex flow patterns, including cardiac flow and abdominal flow imaging. Conventional VFI relies on the use of multiple angles in transmit or receive, or speckle tracking. They are computationally intensive and estimate quality may be sacrificed to improve computational speed. In this

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work, we report a vector flow estimation technique using a deep neural network. The network extracts feature from high-pass filtered beamsummed RF data of two consecutive Doppler packets. For each packet, the RF data have azimuth and axial dimensions. It then performs estimation of vector flow in the feature space, and maps the estimate back to the spatial domain. The total computation time is 0.11 s for a pair of Doppler frames of 1024 × 128 samples. The performance of the method is characterized using Field II simulation studies, flow phantom studies, and an *in vivo* liver study with a Verasonics Vantage 256 scanner. The simulation and flow phantom studies show good agreement between the estimates and the ground truth. The *in vivo* studies demonstrate that the method is capable of characterizing complex flow patterns in human liver vessels.

9:55–10:10 Break

10:10

3aBA8. High-volume-rate ultrasound 3-D flow imaging with a 2-D spiral array: A simulation study. Rebekah Maffett (Elec. and Comput. Eng., Univ. of Waterloo, 250 Laurelwood Dr., Waterloo, ON N2J 0E2, Canada, rebekah.maffett@uwaterloo.ca), Billy Y. Yiu, Adrian J. Chee (Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada), Piero Tortoli (Information Eng., Univ. of Florence, Firenze, Italy), and Alfred C. Yu (Elec. and Comput. Eng., Univ. of Waterloo, Waterloo, ON, Canada)

High-volume-rate ultrasound imaging enables the rendering of complex flow dynamics in 3-D; however, high data rates pose significant hurdles towards real-time implementation. A sparsely populated 256-element 2-D spiral array coupled with unfocused transmissions is a potential solution to these challenges. This work analyzes the impact of sparse element distribution on flow velocity estimation with spherical waves (virtual point source 20mm behind the probe) to extend the field of view beyond the array footprint. Field II was used to simulate straight tube flow (6mm-diameter; 45° tilt; 30cm/s plug-flow; Transmission: 7.5 MHz, 3-cycle) to analyze the velocity estimation bias without the influence of surrounding tissue. It was found that the bias at the inlet, mid-section, and outlet were 20.3%, 8.7%, and 4%, respectively; this variation in bias is due to the change in beam-flow angle from 57.8 deg to 34.9 deg. To evaluate the wide-angle and grating-lobe performance of this configuration, the median Doppler power was compared between the center flow region, flow at the extended regions, and the grating lobe region; the relative power to the center flow region was found to be -12.3dB and -19.5dB for extended and grating-lobe regions respectively. These results suggest that the flow estimation region can be extended beyond the array footprint area but extra considerations on the beam-flow angle must be taken, especially in the case of vector flow.

Invited Papers

10:25

3aBA9. Challenging the multimodal ultrasound with GPU-based research ultrasound platforms. Marcin Lewandowski (us4us Ltd., Pawinskiego 5B, Warsaw 02-106, Poland, marcin@us4us.eu), Mateusz Walczak, and Beata Witek (us4us Ltd., Warsaw, Poland)

Old-school ultrasound was all about discrete modalities (B-mode, Doppler). Today, we prefer to combine multiple methods and advanced processing algorithms to extract more diagnostic data out of the raw RF ultrasound echoes. Thus, high-speed acquisition and processing of channel data have become a standard feature of contemporary research systems. I will present our developments in the programmable and scalable ultrasound systems featuring GPU-based processing. The implementation of multimodal methods is enabled through a high-performance data streaming and software processing approach. We will look into GPU processing and how challenging real-time realization of the Software Defined Ultrasound paradigm can be. We will also discuss the many new opportunities and open-source tools the software approach nevertheless opens up, which can be used for both standard ultrasound processing and advanced post-processing (e.g., machine learning).The case-studies presented show how the platforms can be used as versatile research tools, as well as demonstrators for commercial introduction of the technology. A sneak preview of a 3-D-ready research system will conclude my talk.

10:45

3aBA10. New real-time Doppler developments based on ULA-OP platforms. Piero Tortoli (Information Eng., Università di Firenze, via Santa marta 3, Firenze 50139, Italy, piero.tortoli@unifi.it) and Enrico Boni (Information Eng., Università di Firenze, Firenze, Italy)

The recent introduction of open scanners has substantially boosted the development of new Doppler methods capable of performing detailed measurements of blood flow or tissue velocity. The flexibility of such scanners allows to test novel transmission strategies and to (Doppler) process the related echoes, either in real-time or off-line, depending on the available processing power. In this paper, the main characteristics of the ULA-OP open scanners are briefly reviewed and shown ideal for advanced real-time Doppler imaging. Significant application examples aimed at solving classic limitations of Doppler investigation are then presented. First, coded TX (made possible by the on board linear transmitters) is shown capable of significantly improving the signal-to-noise ratio of blood velocity measurements in deep vessels. Second, the ambiguities due to the unknown Doppler angle are overcome by novel vector Doppler approaches addressed to the investigation of all SVs aligned along one or multiple lines. Finally, ULA-OP is shown suitable to perform, in real-time, plane-wave-based color flow imaging at high-frame rate. This capability is exploited to produce correct flow maps even in presence of rapid blood velocity changes or to improve the quality of color flow images by extending the packet size.

11:05

3aBA11. Using the Doppler effect to characterize microbubble shell viscoelasticity. Outi Supponen (Mech. Eng., Univ. of Colorado, Boulder, CO), Francesco Guidi (Information Eng., Univ. of Florence, Florence, Italy), Awaneesh Upadhyay, Hendrik J. Vos (Biomedical Eng., Erasmus MC, Rotterdam, The Netherlands), Piero Tortoli (Information Eng., Univ. of Florence, Firenze, Italy), and Mark Borden (Mech. Eng., Univ. of Colorado, 1111 Eng. Dr., Campus Box 427, Boulder, CO 80309, mark.borden@colorado.edu)

The experimental testing and quality control of microbubble formulations for ultrasound applications requires high-throughput measurement of physicochemical properties, such as shell elasticity and viscosity. Single-bubble optical experiments are slow and expensive. Ultrasound attenuation experiments on microbubble populations can be done quickly, but the measurement is averaged over the ensemble, and it is often difficult to separate effects of microbubble size from shell viscoelasticity. Here, we discuss a relatively simple alternative method of using the Doppler effect to measure ultrasound radiation force-induced displacements of many individual microbubbles within a population. Our method involves insonifying a diluted suspension of freely floating microbubbles with a linear-array ultrasound probe driven by an open scanner. The microbubble displacements along the axis of the insonified region were acquired from the frequency shifts in the measured echo signals using the multi-gate spectral Doppler approach. These measurements were compared with theoretical peak microbubble displacements computed by combining a modified Rayleigh-Plesset equation and a balance of translational forces. The shell elasticity and viscosity producing the measured peak microbubble displacement was determined by assuming a resonant bubble size at each driving frequency. Our initial results are encouraging, and demonstrate a step toward high-throughput microbubble characterization using the Doppler effect.

11:25

3aBA12. Rational design of flow phantoms for evaluation of new Doppler ultrasound techniques. Adrian J. Chee, Billy Y. Yiu, Chung Kit Ho (Univ. of Waterloo, Waterloo, ON, Canada), and Alfred C. Yu (Univ. of Waterloo, EIT 4125, Waterloo, ON N2L 3G1, Canada, alfred.yu@uwaterloo.ca)

With various advances in Doppler ultrasound technology such as high-frame-rate flow imaging and vector flow estimation, there is a growing need to devise appropriate phantoms that can holistically assess the accuracy of the derived flow estimates. Straight-tube models simply cannot serve this purpose, nor can spinning discs that simulate tissue motion instead of flow dynamics. In this presentation, we shall highlight a series of innovations in devising flow phantoms that can foster meticulous performance evaluation of new Doppler ultrasound techniques. First to be discussed is the design of a novel wall-less spiral phantom with omnidirectional flow (i.e., from 0 to 360 deg). With a three-loop spiral geometry with 4 mm lumen diameter and 5 mm pitch, this phantom was developed using a combination of computer-aided design, 3-D printing of vessel core, and lost-core casting with polyvinyl alcohol cryogel. It is useful for testing the performance of flow vector estimators. The design of anatomically realistic flow phantoms will also be described, and a representative example based on patient-specific aneurysm geometry will be highlighted. In addition, methodological innovations in developing walled arterial phantoms will be presented. Their application will be demonstrated in the context of testing new vascular imaging techniques that simultaneously track wall motion and blood flow.

Contributed Paper

11:45

3aBA13. An anthropomorphic prostatic urinary tract phantom for Doppler-based assessment of voiding dysfunction. Takuro Ishii, Chung Kit Ho, Hassan Nahas, Billy Y. Yiu (Univ. of Waterloo, Waterloo, ON, Canada), Adrian J. Chee (Univ. of Waterloo, 250 Laurelwood Dr., Waterloo, ON N2T 2S1, Canada, adrian.chee@uwaterloo.ca), and Alfred C. Yu (Univ. of Waterloo, Waterloo, ON, Canada)

Direct assessment of urinary flow dynamics during voiding is challenging to perform *in vivo*. New Doppler ultrasound techniques are being developed for this purpose, but their performance needs to be systematically evaluated using phantoms. Here, we present a new design protocol to fabricate a urinary tract phantom with realistic anatomical, mechanical and urodynamic properties. In this protocol, computer-aided design of the urethra geometry was first drafted and it was physically built using 3-D printing. Subsequently, the flexible urinary tract tube was fabricated from polyvinyl alcohol cryogel (Young's modulus: 26.6 kPa), while an agar–gelatin mixture (Young's modulus: 17.4 kPa) forms the surrounding tissue mimic. Finally, the urethra phantom was connected to a flow circuit that simulates voiding. Deformable phantoms were devised for a normal urethra and a diseased urethra with benign prostatic hyperplasia (BPH) obstruction. The phantoms' morphologies were measured with ultrasound imaging and their flow profiles were assessed using Doppler ultrasound. During voiding, short-axis lumen diameter at the obstruction of the BPH-featured phantom was significantly smaller (0.91 vs 2.49 mm) while the maximum flow velocity was higher (59.3 vs 22.7 cm/s) compared to those of the normal urethra. These fabricated phantoms were effective in simulating urethra deformation resulting from urine passage and may promote the development of new Doppler-based urodynamic measurement techniques.

Session 3aCA

Computational Acoustics, Structural Acoustics and Vibration, and Signal Processing in Acoustics: Application of Model Reduction in Computational Acoustics

Kuangcheng Wu, Cochair
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D. Keith Wilson, Cochair
US Army Cold Regions Research Lab., 72 Lyme Rd., Hanover, New Hampshire 03755

Chair's Introduction—8:00

Invited Papers

8:05

3aCA1. Reduced-order models for piezoelectric systems. Jeffrey Cipolla (Appl. Sci., Thornton Tomasetti, Inc., 1825 K St. NW, Ste. 350, Washington, DC 20006, cipolla@wai.com)

Large-scale analysis of transducer arrays usually requires detailed, high-resolution modeling to represent the coupling effects of piezoelectric, elastic, and fluid media. However, reduced-order models, in the form of low-DOF lumped-parameter engineering models remain essential for system design. Here, we show how to bridge the gap between these model classes by deriving a simple means to compute the modes of an arbitrary piezoelectric system. Our approach exploits the intrinsic disparity in time scale between the electric and elastic fields to reduce the coupled problem into an augmented, elastic-only, system of equations. This system is shown to have the same mathematical characteristics as the elastic system, insofar as no new singularities appear. The piezoelectric-elastic modal system is fully general, and compatible with large-scale finite element representations of complex transducer designs. Numerical examples are shown to demonstrate the efficiency and practicality of the modal analysis.

8:25

3aCA2. Orthocomplement reduced order models for strongly coupled structural-acoustic systems. Jeffrey Cipolla (Appl. Sci., Thornton Tomasetti, Inc., 1825 K St. NW, Ste. 350, Washington, DC 20006, cipolla@wai.com)

Despite the enormous strides in computing power, reduced-order models (ROMs) remain essential for design optimization and early stage design. The energetically optimal basis for ROMs of vibroacoustic systems derives from the coupled Eigen problem: no basis of smaller dimension contains a greater amount of system energy. While vital, this L2-type framework is also insufficiently accurate on a pointwise basis for many problems. A simply supported beam with a point load is an obvious thought experiment: the smooth modal functions converge to the correct displaced shape very slowly. The well-known "residual mode" or orthocomplement concept overcomes this problem, by augmenting the space of modal functions with static solutions to the point loads. These new residual modes are orthogonalized against the modal space, and used otherwise conventionally. This work describes the development of ROMs of submerged structures by extending the residual mode approach to the strongly coupled structural-acoustic case. We derive methods to use these ROMs in substructures. Numerical examples show that the structural acoustic residual mode formulation is very effective in improving the solution method in the low-frequency limit, especially in the fluid, and also at the high-frequency limit, where response quantities tend to the average of the direct-solve FEM response.

8:45

3aCA3. Large scale structural acoustics in Sierra-SD. Gregory Bunting (Computational Solid Mech. & Structural Dynam., Sandia National Labs., 709 Palomas Dr. NE, Albuquerque, NM 87108, gbuntin@sandia.gov) and Timothy F. Walsh (Computational Solid Mech. & Structural Dynam., Sandia National Labs., Albuquerque, NM)

In this talk, we will present an overview of structural acoustics and model reduction capabilities in Sierra-SD (Structural-Dynamics). Sierra-SD is a massively parallel finite element application for structural dynamics and acoustics, having recently demonstrated the implicit solution of models with over 2 billion degrees of freedom and running on up to one hundred thousand distributed memory cores. Domain decomposition and MPI parallelism are used to split a large domain into many small domains that can be solved in parallel. Sierra-SD offers a wide range of capabilities for model reduction. For acoustic models, infinite elements and perfectly matched layers are available to reduce domain size. For structural models, Craig-Bampton style superelements are available to reduce specific components in the structural portions of the model. Various coupling and handoff methods between linear structural acoustics codes and nonlinear solid and fluid mechanics codes are used to reduce model size. Large-scale, realistic applications will be presented on models such as ship shock loading and vibration of reentry bodies. [Sandia National Laboratories is a multimission laboratory managed and operated by National Technology and Engineering Solutions of Sandia, LLC., a wholly owned subsidiary of Honeywell International, Inc., for the U.S. Department of Energy's National Nuclear Security Administration under Contract DE-NA-0003525.]

9:05

3aCA4. Reduced order modeling of interior noise for an aircraft during early stage design. Nickolas Vlahopoulos (Univ. of Michigan, 2600 Draper Dr., Ann Arbor, MI 48109, nickvl@umich.edu) and Geng Zhang (Michigan Eng. Services, Ann Arbor, MI)

The design of an aircraft must balance the performance in many disciplines in order to achieve performance expectations at minimum weight and cost. During the conceptual design phase there is no time for developing detailed simulation models and decisions are typically made either by using low fidelity models or existing data and regression models. However, the decisions made during the conceptual design phase greatly affect the performance of the aircraft and the associated cost. The structural design during early product development focuses on strength, fatigue, corrosion, maintenance, inspection, and manufacturing. The structures are designed as a load carrying shell reinforced by frames and longitudinals, and a skin-stringer construction supported by spars and stiffeners. The structural design of the fuselage is directly related with the interior noise performance of the aircraft. Therefore, an easy to use, computational, physics based capability for interior noise prediction, which requires small computational resources, and can operate using the same amount of limited data which is available for the structure during the early product development stages, is a key enabler in bringing interior noise computations in early aircraft design. A case study of such capability will be presented.

9:25

3aCA5. Classical modal analysis and asymptotic modal analysis of structural and acoustic systems. Shung H. Sung (SHS Consulting LLC, 4178 Drexel Dr., Troy, MI 48098, ssung1972@gmail.com) and Donald J. Nefske (Engineering Mechanics Group, LLC, Troy, MI)

The Classical Modal Analysis (CMA) method is commonly used to solve for the vibroacoustic response of large scale structural and acoustic systems. In this method, the modal equations of motion are developed and solved for the modal coefficients to form a modal series solution to predict the structural and acoustic responses. The CMA method generally works well in the low-to-medium frequency range. However, at high frequencies, the large number of modes required in the summation limits the usefulness of the CMA method and alternative methods, such as Statistical Energy Analysis (SEA) and Energy Finite Element Analysis (EFEA) have been developed. As an alternative to SEA and EFEA, the Asymptotic Modal Analysis (AMA) method has been developed in which the modal series summation in the CMA method is asymptotically approximated based on the overall frequency and band averaging of the modal series. The AMA method thereby predicts the band averaged frequency response in the high frequency range and also has the advantage of providing the spatial distribution of the response based on the summed modes in the frequency band. In this paper, an example of coupled cavity-plate system subject to external random white noise pressure load excitation is developed to illustrate the CMA and AMA methods and to address their accuracy.

9:45

3aCA6. Recent developments of faster numerical methods for fluid-structure interaction analysis. Kuangcheng Wu (NSWCCD, 9500 MacArthur Blvd, West Bethesda, MD 20817, kcwu@msn.com), Alyssa Bennett (Univ. of Michigan, West Bethesda, MD), and Xingguang Diperna (NSWCCD, West Bethesda, MD)

Frequency Response Functions (FRF) haves been used to aid noise and vibration designs in various industries. Those design considerations include resonance avoidance, vibration reduction, etc. Numerical methods have been widely applied to predict Frequency Response Functions (FRF) of structures. However, the computational resources (i.e., CPU time, memory, disk space) needed to solve large and detailed numerical models are getting large. Furthermore, the need to resolve resonant response peaks can drive up the number of FRF calculations required. Lately, advanced numerical techniques based on a Krylov subspace and Galerkin Projection (KGP) and Pade Approximation have been demonstrated that they can significantly accelerate the overall process by approximating the frequency dependent response (calculating the forced response at only a few frequencies, then using KGP or Pade approximates to reconstruct the FRF for the rest of desired frequency points.) This paper will present the latest enhancements to the KGP: modeling of viscoelastic material via its complex modulus representation and adaptive capability (AKGP) in automating the frequency sweep process via calculated tolerance error. To illustrate the accuracy and efficiency of the new enhancements, a numerical example will be exercised and used as a benchmark to compare different numerical tools with and without the AKGP.

10:05–10:20 Break

Contributed Papers

10:20

3aCA7. Applying a numerical Green's function approach to scattering within complex seafloor environments. Aaron Gunderson (Appl. Res. Labs., The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, aarong@arlut.utexas.edu)

A numerical approach to solving for Green's functions allows far-field scattering to be solved within variable seafloor environments. The approach is applicable for both seafloor and buried target scattering problems. A robust near-field model is required for near-field solution evaluation and Green's function determination. In this work, a 3-D finite element model is used. Far-field scattering results are determined through numerical integration of the Helmholtz-Kirchhoff surface integral within the near-field of the target or interface. This approach has previously been shown to give accurate results within flat sediment interface environments; now the technique is extended to more complex interface structures, in which the benefit of the numerical Green's function technique is most apparent. For flat interface environments, analytic Green's functions are known, making numerical approaches redundant. For more complex environments, exact solutions are difficult to come by, and analytic approximations grow in error with

increasing environmental complexity. Furthermore, the numerical approach requires only a single field measurement to be made within the near-field model, making the approach computationally efficient and feasible for a wide variety of scattering problems. [Work supported by ONR, Ocean Acoustics.]

10:35

3aCA8. Distributed acoustic sensing of shots in realistic environments. Adrien Dagallier (Acoust., ISL, 5, rue du général Cassagnou, Saint-Louis 68300, France, adrien.dagallier@isl.eu), Sylvain Cheinet (Acoust., ISL, Saint-Louis, France), Daniel Juvé (LMFA, UMR CNRS 5509, Université de Lyon, Ecole Centrale de Lyon, Écully, France), Timothée Surgis, and Thierry Broglin (Acoust., ISL, Saint-Louis, France)

Upon firing, most weapons emit very loud sounds. These sounds propagate over the battlefield and are distorted by the atmospheric effects, absorbed by the ground, reflected on or diffracted around buildings or mountains. It is of obvious operational interest to develop sensing systems to localize these sound sources, with arrays of distributed sensors. This study develops an original sensing approach. It uses the time-matching method, based on finding the best match between pre-calculated times of arrivals (TOAs) of the shot sounds and measured TOAs from a set of synchronous, distributed sensors. Predicting the TOAs requires a physical model able to factor in the impact of complex 3-D environments (wind and sound speed gradients, obstacles), and of complex sound sources (e.g., combination of muzzle blast and supersonic projectile wave). A very fast interface-tracking model is used, based on Sethian's Fast-Marching method, for pre-calculating the TOAs in a general and comprehensive framework. Applications to localization of shots in urban environments and to localization of long range artillery gun are presented. They demonstrate that, compared to standard methods, the above matching-and-marching approach can work without classification, with less sensors, or with a much smaller baseline array.

10:50

3aCA9. Feature reduction through manifold learning for a geospatial model of ambient soundscapes. Katrina Pedersen (Phys. and Astronomy, Brigham Young Univ., Provo, UT, katrina.pedersen@gmail.com), Mark K. Transtrum, Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Shane V. Lympany, Michael M. James, Alexandria R. Salton, and Matthew F. Calton (Blue Ridge Res. and Consulting, Provo, UT)

Manifold learning is a type of nonlinear dimensionality reduction that helps identify a minimal combination of features to characterize data. This presentation discusses the use of diffusion maps, a type of manifold learning, on a dataset of 68 geospatial features which cover the contiguous United States. These geospatial features have been used previously to make predictions of ambient soundscapes through an ensemble of machine learning models. As the current model capabilities are expanded to predict the ambient soundscape outside of the contiguous United States, decisions must be made about which geospatial features are required for accurate models. In particular, some of these 68 features are not available or are expensive to obtain for regions outside of the contiguous United States. Diffusion maps can assist in identifying the features, or combinations of features, that best characterize the data space. [Work supported by a U.S. Army SBIR.]

11:05

3aCA10. An improved Cramer's scheme for computing sound fields in a stratified medium. Jianxiong Feng (Mech. Eng., Purdue Univ., 177 S Russel St., West Lafayette, IN 47907, feng191@purdue.edu) and Kai Ming Li (Mech. Eng., Purdue Univ., West Lafayette, IN)

The Fast-Field Program (FFP) is a powerful numerical tool for calculating sound fields in an inhomogeneous atmosphere. The medium can be divided into a number of layers in the case of a vertical stratification. A linear system of simultaneous equations can be established using the boundary conditions between each layer. To solve the problem, a global matrix is formed and typically be solved by means of the LU decomposition method. However, the inversion of the matrix is time-consuming when the matrix size is large. Since a limited number of layers closed to ground are usually needed in the prediction of the sound fields, hence, the global matrix approach is re-examined in the present paper by seeking a numerically efficient method for matrix inversion. A close examination of the governing equations suggests that a pseudo penta-diagonal matrix can be formed where the associated determinants can be calculated by using their diagonal elements. The computational cost for calculating the kernel function of a specific layer is then linearly proportional to the matrix size. This paper discusses an approach in setting up the penta-diagonal matrix and examine an efficient Cramer scheme in calculating the determinants of the matrix and its co-factors.

11:20

3aCA11. Levin's collocation method for predicting sound fields in the presence of sound speed gradients. Yiming Wang (Purdue Univ., 2120 McCormick Rd., Apt. 711, West Lafayette, IN 47906, wymchihiro@gmail.com) and Kai Ming Li (Purdue Univ., West Lafayette, IN)

Sound propagation in a stratified medium has been studied for decades. The Ray tracing method, which is one of the most popular methods, calculates the ray path and propagating angle of a sound ray. However, the ray method is not accurate under many circumstances compared with the wave-based numerical schemes. The Fast field program (FFP), an efficient method for numerical evaluation of the Fourier integral, is accurate but requires significant computational resources for high frequency sound fields because of the highly oscillatory functions found in the integrand. In this paper, a Levin's collocation method is introduced and used in the evaluation of the sound fields above a locally reacting ground with the presence of the sound speed gradient. The method is far more efficient than most other available numerical techniques such as the FFP and Parabolic Equation method.

11:35

3aCA12. Feature reduction of crowd noise used for machine learning classification. Brooks A. Butler (Brigham Young Univ., 141 S 200 E, Provo, UT 84604, brooks.butler93@gmail.com), Spencer Wadsworth, Dallen Stark, Katrina Pedersen, Blake Forkey, Mylan R. Cook, Eric Todd, Kent L. Gee, and Mark K. Transtrum (Brigham Young Univ., Provo, UT)

This paper discusses methods used to identify the relative importance of audio features in a supervised machine learning (ML) model for predicting crowd behavior at collegiate basketball games. This work builds upon previous research done at Brigham Young University in which ML classifiers were trained using audio recordings of crowds at collegiate basketball games. Previous classifiers were built using several hundred features. A future goal is real-time classification of crowd behavior. This requires reduced computational time for calculating features and classifying audio data. Feature reduction should decrease the computational time for both of these tasks. The audio features can be separated into two categories: (1) spectral features and (2) low-level signal parameters. This paper discusses feature reduction methods—such as random forest Gini importance and p-value feature selection—and compares reduced feature sets. The number of features used in crowd noise classification can be reduced to 10-15 features before seeing a significant decrease in prediction accuracy.

Session 3aED

Education in Acoustics: Hands-On Demonstrations for Middle- and High-School Students

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, Pennsylvania 16802

L. Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Rd., Suite 300, Melville, New York 11747

Acoustics has a long and rich history of physical demonstrations of fundamental (and not so fundamental) acoustics principles and phenomena. In this session, “Hands-On” demonstrations will be set-up for a group of middle- and high-school students from the San Diego area. The goal is to foster curiosity and excitement in science and acoustics at this critical stage in the students’ educational development and is part of the larger “Listen Up” education outreach effort by the ASA. Each station will be manned by an experienced acoustician who will help the students understand the principle being illustrated in each demo. Any acousticians wanting to participate in this fun event should e-mail Keeta Jones (kjones@acousticalsociety.org).

Chair’s Introduction—8:30

Invited Paper

8:35

3aED1. Using Wikipedia to promote acoustics knowledge for the International Year of Sound 2020. William J. Murphy (Noise and BioAcoust. Team, Div. of Field Studies and Eng., National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov), Thais C. Morata (Noise and BioAcoust. Team, Div. of Field Studies and Eng., National Inst. for Occupational Safety and Health, Cincinnati, OH), and John Sadowski (Office of the Director, National Inst. for Occupational Safety and Health, Washington, DC)

Wikipedia is an important mechanism to share science information with the public. The National Institute for Occupational Safety and Health (NIOSH) adopted a multi-component strategy to improve the health content on Wikipedia. For the International Year of Sound (IYS 2020), the Acoustical Society of America and NIOSH organized the online event Wiki4YearOfSound2020, to facilitate the improvement of Wikipedia content related to acoustics. IYS 2020 is a global initiative to highlight the importance of sound in all aspects of life and improve our understanding of sound-related issues at an international level. The Wiki4YearOfSound2020 platform (available in several languages) provides guidance to all who join. Ways to participate include promoting the project, suggesting a topic that should have a Wikipedia article, improving or translating existing articles, articles, or writing new ones. Educational programs can count on a platform and tools to train students to contribute high-quality content to Wikipedia as part of their class assignments. Throughout 2020 the Wikimedia outreach dashboard will allows anyone to monitor contributions and reach. This campaign is an opportunity to make content in acoustics one of the better-developed areas within Wikipedia, while providing quality information to everyone in the world-where they are *actually* looking for it.

Session 3aMU

Musical Acoustics: General Topics in Musical Acoustics I

Andrew A. Piacsek, Chair
Physics, Central Washington University, 400 E. University Way, Ellensburg, Washington 98926

Contributed Papers

9:00

3aMU1. A tone tutor for training student musicians. Gerry Ruch (Phys., Univ. of St. Thomas, 2115 Summit Ave., M.S. OWS 153, St. Paul, MN 55105, ruch6633@stthomas.edu) and Chris Kachian (Music, Univ. of St. Thomas, St. Paul, MN)

We present a real time tool for guitar students to practice the production of good tone. Empirically, according to an informal survey of guitar teachers, warm tones are generally preferable to bright tones. Our tool quantifies the tone by determining the fundamental frequency of the note and examining the harmonics. A note with many strong harmonics well above the fundamental sounds bright while a note with only harmonics near the fundamental sounds warm. We have built a smartphone app that calculates the harmonic average of an incoming signal and presents the resulting measurement in real time. This app can be used by a music student during solo practice sessions between lessons to develop good tone on their instrument.

9:15

3aMU2. Atom music: exploring the atomic world through sound. Jill A. Linz (Phys., Skidmore College, 815 N. Broadway, Saratoga Springs, NY 12866, jlinz@skidmore.edu)

Atom music is a fun and educational way to investigate the sound of the atomic world using music synthesis techniques to create atom "songs." Students gain insight into the structure of sound and its relation to the structure of matter using Fourier analysis. By correlating the bright lines appearing in atomic spectra to audible tones, individual atom notes can be identified. These notes then comprise a musical scale that is unique to each element. From these scales, atom songs can be created using programs such as Audacity, Logic or Max. Specific atomic traits can be incorporated that allow for creativity in the making of an atom song. Digital synthesizers let the user create waveforms to any specification and allow students to explore the atomic world by creating their music. Simple molecules can be mimicked by combining the music of different atoms to form atom "bands." More advanced students can use signal processing methods to gain deeper insight into the atom's sound. Examples of how atom music has been used can be found in a recent article published online by Physics Today titled *Composing Atom Music*. This presentation explores some of the methods and software used to create atom music.

9:30

3aMU3. Mathematical music theory and the representation of Igbo music. Andrea Calilhanna (MARCS Inst. for Brain, Behaviour and Development, 2 Kayla Way, Cherrybrook, Sydney, New South Wales 2126, Australia, A.Calilhanna@westernsydney.edu.au) and Stephen G. Onwubiko (Univ. of Nigeria, Enugu, Nsukka, Nigeria)

The purpose of our paper is to introduce African mathematical music theory: to analyze and represent African (Nigerian) music through visualizations and sonifications of beat-class theory. Many of the current analytical methods to study African (Nigerian) music lack the qualities necessary to recognize the intricate imbibements from music. Western music theory has evolved largely as a result of non-Western influences and beat-class theory provides a viable solution for the lapses in analytical work which have led to the superimposition of Western ideas of Nigerian music. Musical set theory uses the universal language of mathematics to explore the complexity, beauty and cultural significance of Nigerian music. In our paper, we will provide a discussion of meter in Ikpirikpi Ogu, the Ohafia War Dance of Igboland of Nigeria, to explore how Igbo music from Nigeria can be analyzed through the narrative of mathematical music theory using visualizations and sonifications of beat-class theory in circular cyclic graphs and ski-hill graphs to represent and acknowledge the listener's psychoacoustic experience of music.

9:45

3aMU4. Decolonizing African music with visualizations and sonifications using beat-class theory. Andrea Calilhanna (MARCS Inst. for Brain, Behaviour and Development, 2 Kayla Way, Cherrybrook, Sydney, New South Wales 2126, Australia, A.Calilhanna@westernsydney.edu.au) and Stephen G. Onwubiko (Univ. of Nigeria, Enugu, Nsukka, Nigeria)

In a recent paper, we introduced mathematical music theory as a suitable new approach to analyze the music of Nigeria (Igbo) (Calilhanna *et al.*, 2019). This paper revisits those materials and extends the ideas to further discuss why Nigerian music can be studied more suitably using mathematical music theory. We address a key issue facing music curriculum development: the perpetuation of traditional Western music theory to analyze and understand all music around the world. Our paper aims to demonstrate how Western music theory is inadequate for analyzing Nigerian music and we present some ways Western music theory has evolved. We explain how new music theory combines understandings of both music and mathematics and with a focus on musical meter we present a student-centred approach to learning. We achieve this through visualizations and sonifications of beat-class theory, using ski-hill graphs and circular cyclic graphs. This approach enables the listener to accurately articulate their subjective embodied psychoacoustic experience of music, meter and mathematics and incorporates research in neuroscience, music education, music psychology and music acoustics, which consistently indicates the importance of music for the well-being of the individual, school, workplace, and community.

10:00–10:15 Break

10:15

3aMU5. The natural variability of spruce soundboards: Perceptual differences in the in the tonal quality of acoustic guitars. Sebastian Merchel (TU Dresden, Helmholzstr. 18, Dresden 01069, Germany, sebastian.merchel@tu-dresden.de), M. Ercan Altinsoy (TU Dresden, Dresden, Germany), and David Olson (Pacific Rim Tonewoods, Concrete, WA)

The wood of the spruce tree (*Picea spp.*) has been valued for centuries as an ideal soundboard for stringed instruments, due to its material acoustic properties. There is large variability in these properties between individual trees of the same species, and even within an individual log. It stands to reason that this variability would produce audible differences in sound quality in otherwise identical musical instruments. Further, there may be a specific

combination of material properties of the soundboard that would result in optimal sound quality for a given design, as measured by expert listeners. Nine steel string guitars of the same model were produced by the Taylor Guitar Company, with strict control of all production parameters. The guitars varied only in two parameters: the density and the Young's modulus of the soundboard and bracewood. A short music sequence was used for pairwise comparisons in a double-blind listening test. The results suggest that, for this particular model (the Taylor 814ce Grand Auditorium), low density and Young's modulus of the soundboard and bracewood have a positive impact on overall preference. More generally, these results underscore the importance of integrating a given design with the physical characteristics of the component wood.

10:30

3aMU6. A comparison of nonlinear modal synthesis using a time varying linear approximation and direct computation. Mark Rau (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., 660 Lomita Court, Stanford, CA 94305, mrau@ccrma.stanford.edu) and Julius O. Smith (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., Stanford, CA)

Modal synthesis is a common technique to model linear musical systems efficiently using a parallel bank of second-order digital filters to represent the modes of a resonating instrument. However, many instruments exhibit nonlinear behavior when driven at high amplitudes, often resulting in amplitude dependent mode center frequencies. This behavior can be modeled in a simplified manner using a time varying linear modal structure or could be directly modeled using a nonlinear mode approximation such as that of a Duffing oscillator. The time-varying linear approach is compared to the direct approach to gauge if the approximation is valid within perceptual limits. Particular emphasis is put on weakly nonlinear modes which may be found in a string instrument with thin vibrating plates.

10:45

3aMU7. The "dark" energy in-between sonic partials: Parametric estimation and analysis of weak spectral components for musical sound analysis and synthesis. Angela C. Kihiko (Spelman College, 350 Spelman Ln. SW, Atlanta, GA 30314, akihiko@scmail.spelman.edu), Mitsunori Ogihara (Dept. Comput. Sci., Univ. of Miami, Coral Gables, FL), Gang Ren (Ctr. for Computational Sci., Univ. of Miami, Coral Gables, FL), and James W. Beauchamp (Dept. of Elec. and Comput. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Most musical sounds show strong energy concentrations, such as regular time-frequency patterns of sonic partials, as a sparse energy distribution across the spectrographic analysis plane. Most studies of musical sound are based on strong spectral components such as the energy distribution patterns of the harmonic partials. However, between any two sonic partials most musical sounds show interesting low-energy signal patterns such as wide-band articulation or other performance noise. The authors present a musical signal analysis framework that extracts these low-energy spectrographic components by detecting and masking-off the strong sonic partials. A parametric estimation and analysis framework is implemented for an in-depth study of the timbral properties of these weak signal components. Based on the parametric models of several instrument categories, the authors implemented several resynthesis methods for these weak signal components.

11:00

3aMU8. Effect of hammershank material properties on piano struck string sound. Jacob Amero (CETA, Univ. of Hartford, West Hartford, CT), Mario Riccardi (CETA, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, riccardi@hartford.edu), and Philip P. Faraci (CETA, Univ. of Hartford, West Hartford, CT)

The purpose of this study was to investigate the effects of the hammershank material properties of a piano action assembly on the sound produced by the struck string. Using a standard birch hammershank as a baseline, a CAD drawing was developed from which several test hammershanks were fabricated. Alternate woods such as poplar and red oak, as well as several different 3-D printed filament materials were used. Vibrational frequency response data from 0 to 1600 Hz was collected from each of the hammershanks using a vibration exciter and accelerometers. Decay measurements of the first ten harmonics of the struck string were taken from note C40 (f_1 = 262 Hz) on a typical working grand piano outfitted with the baseline and the test samples. Audio recordings of the string decay produced by each sample were also taken for audible comparison. Results will be discussed.

Session 3aNS

Noise and ASA Committee on Standards: Community Noise

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

David S. Woolworth, Cochair
Roland, Woolworth & Associates, 356 CR 102, Oxford, Mississippi 38655

Invited Papers

7:55

3aNS1. A draft model community noise ordinance. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1137, les@nonoise.org)

Once upon a time, it was hoped that the sound level meter would solve all noise pollution problems for communities with clear objective criteria for acceptable and unacceptable noise. That hope failed to be realized because the volume of noise complaints is overwhelming. In the real world, the police cannot respond with a sound level meter to each of the millions of noise calls they receive each year. Yet clearly, decibel levels and sound level meters are critical to noise regulation. This dilemma has stumped the ANSI Model Community Noise Ordinance Working Group for more than 20 years. To avoid this dilemma, this paper proposes a model noise ordinance developed so that police departments can respond to millions of calls each year due to noise, using not just one, but a number of regulatory tools, including (1) nuisance laws, (2) decibel based regulations, (3) operational restrictions (such as setbacks, time of day restrictions, or restrictions based on zoning district), (4) prohibitions (such as no non-emergency honking, no jake brakes, no radios on buses or beaches), (5) equipment requirements (such as muffers or the EPA stamp on motorcycle mufflers), and (6) the plainly audible standard at a specific distance.

8:15

3aNS2. Community sound levels associated with Allianz Field. David Braslau (David Braslau Assoc., Inc., 6603 Queen Ave. S, Ste. M, Richfield, MN, MN 55423, david@braslau.com)

Allianz Field, a 20 000 seat outdoor stadium in St. Paul, MN is the new home of the Minnesota United Loons MLS team. At the start of the project, community ambient levels were initially measured near three residential areas for the environmental review process. Although exposed to noise from a nearby interstate highway and arterial roadways, sound levels were within Minnesota noise standards. Following completion of the stadium, updated background readings were taken to reflect changes in three years since the initial ambient levels were measured. Sound levels were then monitored during the first night game at the venue. Readings showed that crowd level barely exceeded ambient levels and were below the Minnesota standards. Stadium sound system levels were not audible. An evaluation of the potential noise impacts from predicted concert noise was then prepared. While a database of maximum mix levels at 100 feet from various groups was available, these values had to be converted to L10 levels governed by the state standards. Examination of the time history data for selected events provided needed conversion from mix peak to L10. Concert levels projected at the three original monitoring locations show some initial standard violations. Calculating backward in the prediction process showed that concerts with a mix level of 90 dBA would be in compliance with daytime standards while a mix limit of 80 dBA would suffice for late evening performances.

8:35

3aNS3. Community noise exposure and annoyance: A multilevel description. D. K. Wilson (Cold Regions Res. and Eng. Lab., U.S. Army Eng. Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil)

An overview is provided of multilevel modeling of community noise annoyance. By multilevel, it is meant that noise exposure and noise tolerance vary on the community level, and among individuals within the communities. Although these assumptions are rather simple, they lead to a rigorous statistical model with characteristics consistent with noise survey data. Regression analyses with a multilevel, generalized linear model (GLM) enable the model parameters and their variations at the community and individual levels to be distinguished and quantified. Based on a meta-analysis of transportation noise, the community-level and individual-level noise tolerance variances are both evident, although the individual-level variance is stronger. The significance of community-level variance is consistent with the recently introduced concept of community tolerance level (CTL). Although the multilevel model describes and quantifies the community-level variations, it does not by itself explain why communities differ, or how to account for these differences in regulations. Future research should address the relationships between community tolerance variations and factors such as soundscape disturbance and attitude toward the most prominent noise sources.

8:55

3aNS4. Selecting a parametric probability density function for urban sound. Matthew J. Kamrath (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, Matthew.J.Kamrath@erdc.dren.mil), D. Keith Wilson, Carl R. Hart, Daniel J. Breton, and Caitlin E. Haedrich (U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

In an urban setting, sound levels vary over time and space due to transportation, construction, and other community noise sources. Parametric probability density functions (PDFs) can concisely characterize these variations, but the literature does not identify an appropriate PDF that both has a firm theoretical foundation and fits urban sound data well. A Gaussian distribution, which physically corresponds to a single dominant source, sometimes describes a distribution of levels well, but often it does not. Frequently, the distribution falls somewhere between the idealizations of a single dominant source and many comparable sources, so a model that can approximate both cases could perform better than a Gaussian distribution. To that end, this presentation considers the generalized gamma and compound gamma distributions for modeling the normalized mean squared pressure. Creating histograms of acoustic data, which were collected in Boston, provides a basis to compare the distributions using the Kullback-Leibler (KL) divergence. In general, compared to the generalized gamma and lognormal distributions, the compound gamma distribution has a lower KL divergence and thus more closely matches the experimental distributions.

9:10

3aNS5. Evaluating project-related noise impacts on sleep in environmental assessments: A proposal to apply decibel adjustments to nighttime noise events based on their time of occurrence over the sleep period time. David Michaud (Health Canada, Canadian Federal Government, 775 Brookfield Rd., Ottawa, ON J9H7J9, Canada, david.michaud@hc-sc.gc.ca), Stephen Keith, and Allison Denning (Health Canada, Canadian Federal Government, Ottawa, ON, Canada)

Health Canada is updating its Guidance for evaluating noise in environmental assessments. Current advice on sleep is drawn from the World Health Organization (WHO) which advises that average noise within the bedroom not exceed 30 dBA, and noise events not exceed 45 dB LAmax on more than 10–15 occasions. Both limits increase by 15 dB outdoors assuming partially opened windows. The WHO's recommended outdoor nighttime annual average of 40 dBA is also used. Two situations commonly arise: (1) the outdoor annual average sound level is exceeded at baseline; or (2) the number of events during sleep exceeds 15, but are just below threshold. Under these situations, the estimated prevalence of high noise annoyance is applied to protect sleep because it includes a 10 dB nighttime penalty. However, this may not account for the variation in the acoustic threshold that has been observed throughout the sleep period. This paper presents two proposals: (1) apply a decibel adjustment to noise events occurring during periods where thresholds are known to be lower (i.e., the first hour and final 3 h) and/or (2) derive a rating level for estimating annoyance using an alternative nighttime adjustment that splits the night into distinct time periods.

9:25

3aNS6. Population density and community noise in South Florida. Samuel Shroyer (Edward Dugger + Assoc., P.A., 1239 SE Indian St., Ste. 103, Stuart, FL 34997, sam@edplusa.com) and Edward Dugger (Edward Dugger + Assoc., P.A., STUART, FL)

The Miami-Fort Lauderdale-West Palm Beach metropolitan statistical area (MSA) is the eighth most populous in the U.S. but ranks sixteenth in land area. Confined within the twenty-miles between the Atlantic Ocean and the Everglades, most of the land area the MSA envelops cannot expand as its population grows. As a result, continued development has increased the proximity of urban population centers to less-populated "rural" areas. The results of long-term sound level measurements conducted at over 50 locations throughout the MSA are assessed in conjunction with land uses, population densities, and other statistical data to determine conformance with ASA, ANSI, and WHO guidelines and the U.S. EPA prediction model. Initial analyses revealed noise levels at most measurement locations to be inconsistent with these criteria, even in less-populated areas farther from urban centers. The datasets are compared to establish relationships at the local level and are combined based on their land use designations and respective location within municipal and county boundaries, the MSA, and smaller statistical areas for further correlation with standardized guidelines.

9:40

3aNS7. Unintended consequences of some jurisdictional noise ordinances. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

Since the U. S. Environmental Protection Agency published the Noise Control Act of 1972, many jurisdictions have enacted noise ordinances in various forms to protect the public health and welfare. Over time, research and technology have improved both the understanding of noise effects and the capabilities of noise measurement equipment. It follows then that modifications to noise ordinances would incorporate these improvements. However, in seeking to update noise ordinances, it is important to carefully examine the actual ramifications of the wording, practices and limitations being proposed. Words and wording matter a great deal in municipal noise codes. For example, the use of the word "maximum" is often used differently in noise codes than it is understood in acoustical instrumentation terminology. Another example is the difference between the phrases "shall not exceed" the code noise limit and "shall not cause the noise to exceed" the code noise limit. And finally, some ordinances provide additional adjustments to the noise code whenever the ambient noise is greater than the standard limits of the noise code. These examples, and others, can introduce certain unintended consequences in attempts to achieve project noise code compliance. This paper examines these code compliance challenges and possible remedies.

9:55–10:10 Break

Contributed Paper

10:10

3aNS8. Comparing two approaches for outdoor ambient noise level measurements. Zachary T. Jones (Phys. and Astronomy, Brigham Young Univ., 580 Wymount Terrace, Provo, UT 84604, ztj1@sbcglobal.net), Mylan R. Cook, Kent L. Gee, Mark K. Transtrum (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Matthew F. Calton (Blue Ridge Res. and Consulting, Provo, UT), and Michael M. James (Blue Ridge Res. and Consulting, Asheville, NC)

In ambient community noise measurements, two possible setups involve an elevated microphone or on the ground [ANSI 12.9-2005/Part 2]. This study compares the elevated and ground-based microphone measurement approaches, particularly for low-frequency wind noise contamination. Wind speed measurements were made at heights of 0.15 m and 1.5 m, and one-third octave spectral histories were recorded simultaneously at the ground and at 1.5 m. The acoustical measurements both used Larson Davis 831C sound level meters, but the elevated microphone setup employed a commercial outdoor windscreen, whereas the ground-based setup utilized a larger reticulated foam windscreen. The measurements, each with durations between 1.5 and 4 h, were repeated at several locations. The results show the ground plate setup is preferable for obtaining wideband spectral measurements for two reasons. First, the wind speed is lower near the ground. Second, for the same wind speed, the larger windscreen yields superior wind noise rejection. These observations are quantified and discussed, and connected to expanding the available noise database for a machine-learning model for soundscape prediction. [Work supported by a U.S. Army SBIR.]

Invited Papers

10:25

3aNS9. Low frequency sounds from entertainment and exercise venues: Methods for prediction, measurement, and annoyance assessment. David S. Woolworth (Roland, Woolworth & Assoc., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com)

The advent of inexpensive low frequency reproduction has made the subwoofer loudspeaker ubiquitous and has in turn driven an increase in low frequency content in many forms of popular music. The result is music played in entertainment and exercise venues contains a lot of low frequency information at high sound levels, while prediction for design, ANSI testing standards, and U.S. ordinances do not properly address the full frequency spectrum in question or the pulsing nature of low frequencies. Furthermore the proliferation of lightweight mixed use construction and development of entertainment districts to revitalize urban areas puts these low frequency sources adjacent to or near residences, creating the need for a more comprehensive set of methods for assessment. This paper provides a compiled set of methods and techniques to assess low frequencies from entertainment and fitness venues, drawing on various standards, industrial noise methods, and the author's experiences and sample data.

10:45

3aNS10. Electrical substation noise control case studies and lessons learned. Joseph A. Keefe (Ostergaard Acoust. Assoc., 200 Executive Dr., Ste. 350, West Orange, NJ 07052, jkeefe@acousticalconsultant.com)

Electrical substation noise comprises noise from electrical transformers (magnetostrictive hum and cooling fan noise) as well as HVAC systems associated with buildings and enclosures. These noise sources are necessary to serve the public need for power distribution. Site sound emissions should have minimal acoustical impact on receptors and in most cases are required to comply with regulatory limits pertaining to noise. A few case studies are presented that address challenging noise control situations for electrical transformers and other electrical substation noise sources, including the acoustical consultant's role in the project, identification of regulatory limits and project criteria, acoustical analysis, approach to noise control recommendations, and lessons learned.

11:05

3aNS11. Community-based infrasound investigation. David Braslau (David Braslau Assoc., Inc., 6603 Queen Ave. S, Ste. M, Richfield, MN, MN 55423, david@braslau.com)

Infrasound is not only associated with wind turbines and volcanoes. Low levels of disturbing infrasound that can have long term adverse impact on residents are also associated with sources such as bridges, transportation, barges, mechanical equipment and entertainment sound systems. This paper describes measurements to identify potential infrasound sources impacting a residence in a Mississippi River town in Minnesota. Initial monitoring was performed in a second floor room over a 24 h period with five samples up to 30 min using an Apollo sound analyzer. Supplemented with sound level meter readings, an average peak at 12.5 Hz appears to be associated with a nearby river crossing bridge. Peaks at this frequency peaks can not be attributed to room dimensions. Subsequent readings were taken in several rooms and outside with a micro-barometer system. These data supported earlier readings but showed differences between rooms and indoor and outdoor spectra. Unfortunately there are no standards or guidelines for indoor infrasound levels and only limited solutions for addressing these community infrasound sources. Whether or how infrasound could be incorporated into community noise ordinances that could possibly prevent adverse residential exposure to infrasound will be discussed.

11:25

3aNS12. Unusual case studies—Investigating outdoor and indoor sources of community noise. Joseph F. Bridger (Stewart Acoust. Consultants, 7330 Chapel Hill Rd.,Ste. 101, Raleigh, NC 27607, joe@sacnc.com), Noral D. Stewart, John Gagliardi, Siddharth Mahajan, and Fred Schafer (Stewart Acoust. Consultants, Raleigh, NC)

There has been much research on problems related to noise in the community. Most of this has concentrated on sources related to transportation, military systems, and ventilation systems. That is because these sources are widespread, and there have been readily available mechanisms to fund the research. Most of this research has concentrated on establishing acceptable quantities of sound and reducing sound to those levels for these common sources. Noise ordinances, if not obsolete or carelessly written, may still fail to recognize some of the unusual noise sources as a problem which could cause nuisance in the nearby community. These sources may be at a great distance, not directly visible or even indoors which requires the use of creative or advanced investigation methods to properly identify and mitigate. The authors present some of the unusual projects encountered by their consulting firm in the recent past and how they approached them.

Contributed Paper

11:45

3aNS13. Study of metadiffusers in broadband improvement of noise barrier performance. Prachee Priyadarshinee (Mech. Eng., National Univ. of Singapore, #21-256B, 38 College Ave. East, Singapore 138601, Singapore, a0135595@u.nus.edu)

In previous studies, installation of Schroeder diffusers on top of noise barriers has been shown to improve effectiveness of noise barriers. However, since these diffusers have thickness of the order of design wavelength, they become very bulky, costly and less practical to use. Their use for lower frequency applications is thus limited. Metadiffusers are subwavelength diffusers, capable of creating similar diffusion characteristics, but their thickness can be 10 to 20 times lower than conventional diffusers. In addition, the absorption performance can be improved by incorporating coupled Helmholtz resonators to further increase noise barrier insertion loss. The reduced thickness and incorporation of Helmholtz resonators enables better performance for a broader range of frequencies, which was not possible with a conventional sound diffuser. It was observed that both 1-D and 2-D metadiffuser designs installed on top of noise barriers improve the performance upto 5dBA in the shadow zone. A combination of number sequences such as maximum length, quadratic residue, primitive root, Luke, ternary, etc. with Helmholtz resonators tuned to different frequencies has also been reported in this study.

Session 3aPA

Physical Acoustics and Engineering Acoustics: Non-Reciprocal and Topological Acoustics

Yun Jing, Cochair
North Carolina State University, 911 Oval Dr., EB III, Campus Box 7910, Raleigh, North Carolina 27695

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, Texas 78712

Chair's Introduction—9:15

Invited Papers

9:20

3aPA1. Topological phononic crystals. Zhengyou Liu (Phys., School of Phys. and Technol., Wuhan Univ., Wuhan, Hubei 430072, China, zyliu@whu.edu.cn)

Valley phononic crystals (VPCs) and Weyl phononic crystals (WPCs) are two typical types of topological phononic crystals. The valleys are the pair of energy extrema at two inequivalent corners of the reduced Brillouin zone of the 2-D hexagonal materials. The valley pseudospin, as new degree of freedom in addition to charge and spin, may provide a great alternative to be used in information encoding and processing. VPCs are 2-D acoustic or elastic artificial materials with valleys, and chiral valley states and robust valley edge transport, can be exhibited in either the acoustic or elastic VPCs. Weyl semimetals are materials in which the electrons have linear dispersions in all directions while are doubly degenerate at single points, called the Weyl points, near the Fermi surface in 3-D momentum space. Weyl points also exist in 3-D phononic crystals for acoustic or elastic waves, referred to as WPCs. The surface arc states in the WPC, not only manifest the normal refraction, but also the negative refraction when turning from one surface to another. In either case, they are immune against reflection. Topological phononic crystals extend topological physics from microscopic scales to macroscopic scales, which may accelerate the practical application of the topological physics.

9:40

3aPA2. Acoustic higher-order topological insulators. Baile Zhang (School of Physical and Mathematical Sci., Nanyang Technolog. Univ., SPMS-PAP-0506, 21 Nanyang Link, Singapore 637371, Singapore, blzhang@ntu.edu.sg)

The concept of topological insulators has recently been transferred into acoustics by designing and constructing acoustic analogues of topological insulators, which nowadays are widely called as "acoustic topological insulators." As dictated by the conventional bulk-boundary correspondence principle, the topologically nontrivial band structure in a d-dimensional topological insulator shall support (d – 1)-dimensional boundary states. However, recent theories have proposed a new class of higher-order topological insulators which do not satisfy the conventional bulk-boundary correspondence principle. For example, a two-dimensional (2-D) second-order topological insulator does not support one-dimensional (1-D) topological edge states, but has topologically protected zero-dimensional (0D) corner states. Three-dimensional (3-D) second-order and third-order topological insulators do not have 2-D topological surface states, but host 1-D hinge states on edges, and 0D corner states on corners. Here we introduce our recent results of designing and constructing acoustic higher-order topological insulators. Both second-order and third-order topological insulators are discussed in a platform of acoustic metamaterials. By direct acoustic measurement, we demonstrate the acoustic bandgap and the in-gap corner states and hinge states.

10:00

3aPA3. Acoustic nonreciprocal topological insulators. Bin Liang (Inst. of Acoust., Nanjing Univ., 22 Hankou Rd., Nanjing, Jiangsu 210093, China, liangbin@nju.edu.cn), Xuefeng Zhu (Huazhong Univ. of Sci. and Technol., Wuhan, China), Jianchun Cheng (Inst. of Acoust., Nanjing Univ., Nanjing, China), Yujiang Ding, Yugui Peng (Huazhong Univ. of Sci. and Technol., Wuhan, China), and Yifan Zhu (Inst. of Acoust., Nanjing Univ., P. R. China, Nanjing, China)

Nonreciprocal acoustic systems, in which the reciprocity obeyed by wave motion in conventional propagation media, offer the possibility to achieve asymmetric transmission and contain rich physics. The emergence of "acoustic rectifier" for the first time realized both theoretically and experimentally the non-reciprocal wave phenomenon that the acoustic energy flows only in one direction and thereby created a "one-way street" for sound. With the extraordinary capability to redefining the common paradigm of wave propagation, nonreciprocal materials are promising for the development of acoustic topological insulator. Later it is proven that giant acoustic nonreciprocity can be provided by an angular-momentum-biased circulator. However, the practical implementation remains challenging due to other difficulties such as nonsynchronous rotation and flow instabilities. The experimental realization of an acoustic Chern insulator is recently reported based on a mechanism that uses an angular-momentum-biased resonator array to break reciprocity, which reduces the required rotation speed by leveraging high Q-factor resonance. Experimental results show that such systems support one-way nonreciprocal transport of sound at its edges, in direct analogy to electronic quantum Hall insulator. The realization of acoustic nonreciprocal insulators opens up opportunities for exploring unique observable topological phases and developing nonreciprocal devices in acoustics, with potential application in diverse scenarios.

10:20–10:35 Break

10:35

3aPA4. Topological edge transport and one-way field localization in acoustic systems. Xue-Feng Zhu (Phys., Huazhong Univ. of Sci. and Technol., Hongshan, Luoyu Rd., No. 1037, Wuhan, Hubei 430074, China, xfzhu@hust.edu.cn)

Recently, the fields of valley acoustics and nonreciprocal acoustics have become hotspots due to the potentials in developing various acoustic devices. In this presentation, we first introduce the concept of programmable routing of topological sound transport through boundary engineering, which reveals the inherent relation between the field symmetries of valley states and structural symmetries of sonic crystals. Three functional devices are exemplified, which are single-crystal-based topological delay-line filter, delay-line switcher and beam splitter. Our results clearly demonstrate the high-transmission valley transport along the folded boundaries, where reflection or scattering is prohibited at the sharp bending or corners due to topological protection. Then we show an analog of stimulated adiabatic passage in acoustic systems, where the cavities and time-varying couplings mimic discrete states and radiation pulses, respectively. With appropriate arrangements of coupling actions, acoustic waves can be efficiently transferred from the initial excited cavity to the target cavity in the forward direction, immune to the intermediate dark cavity. Whereas for the backward propagation, the acoustic energy is perfectly localized in the intermediate dark cavity and completely dissipated. We analytically, numerically, and experimentally demonstrate such unidirectional sound localization and unveil the essential role of zero-eigenvalue eigenstates in the adiabatic passage process.

10:55

3aPA5. Transmisison hysteresis and edge modes in bounded space-time composites. Hussein Nassar (Univ. of Missouri - Columbia, Lafferre Hall, Columbia, MO 65201, nassarh@missouri.edu)

The vibrational frequency response of bounded composites, e.g., metamaterials and sonic crystals, is often understood thanks to band diagrams established in the absence of boundaries. Introducing a pump wave that modulates in time the properties of the composite challenges the correspondence between the vibrations picture and the waves picture. The talk revisits this correspondence in the context of the nonreciprocal acoustics of space-time composites. Specifically, we establish in the weak coupling regime how the hybridization of total bandgaps into pairs of one-way bandgaps triggers nonreciprocal hysteresis transmission loops in the space-frequency domain and alters, qualitatively and quantitatively, the vibrational frequency response in the presence of reflecting boundaries. The theoretical analysis is assessed numerically and exploited to shed new light on previously obtained experimental data. Last, extrapolating our study to the strong coupling regime, transmission hysteresis is shown to explain the emergence of topological vibrational modes with one-way edge-bulk and bulk-edge transitions.

11:15

3aPA6. Dynamics of space- and time-modulated metastructures. Massimo Ruzzene (Mech. Eng., Univ. of Colorado Boulder, 1111 Eng. Dr., Boulder, CO 80309, massimo.ruzzene@colorado.edu)

Topology has recently emerged as a principle governing unique wave transport phenomena through interface or edge modes that are impurity-immune and potentially unidirectional. In this context, this presentation illustrates the effect on dispersion topology of modulations of properties in space and time. The two modulation strategies are shown to lead to dual phenomena, which can be described by interchanging frequency and wavenumber, and that manifest themselves as gaps enabling frequency/wavenumber-selective wave manipulation. Time modulation produces narrowband tunable reflection at a frequency determined by the modulation, which is illustrated on a beam waveguide with switching negative impedance piezoelectric shunts. Adiabatic, or slow, space modulation of the stiffness properties of a plate structure drives the transition from a localized state at one boundary, to a bulk state and, finally, to another localized state at the opposite boundary. The presented investigations suggest the application of modulation strategies for single-port tunable filtering devices that may be implemented in acoustic, mechanical or photonic platforms. Moreover, the considered experimental platforms allow the exploration of various unique properties associated with time and/or space modulation, including filtering, frequency conversion, non-reciprocity, and topological pumping for wave redirection.

Contributed Paper

11:35

3aPA7. Standing waves in structures with spatiotemporally modulated material properties. Benjamin M. Goldsberry (Appl. Res. Labs., 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 and elastic wave metamaterials with time- and space-dependent material properties have received great attention as a means to increase the degree of control over the propagation of linear waves through a medium. Previous works have shown that propagating waves in a modulated medium display non-reciprocity by means of asymmetric frequency and wavenumber conversion between two counter-propagating modes [K. Yi *et al.*, *Phys. Rev. B* **96**, 104110 (2017)]. In the present study, we investigate standing acoustic waves in a finite medium with time- and space-dependent material properties. A semi-analytical approach based on coupled mode theory is derived, as well as a finite element approach that can consider more complex geometries. The effects of space-only, time-only, and space-time modulations of the material properties on standing waves in a finite system are explored. The present analysis leads to potential applications in acoustic communications, vibration suppression, and energy harvesting. [Work supported by NSF EFRI.]

Session 3aPP

Psychological and Physiological Acoustics: Technological Advancements in Hearing Research (Poster Session)

Tian Zhao, Chair
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All posters will be on display from 9:00 a.m. to 11:00 a.m.. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 to 10:00 a.m. and authors of even-numbered papers will be at their posters from 10:00 a.m. to 11:00 a.m.

Contributed Papers

3aPP1. Auxlab—An integrative open-source computing environment software for audio and speech signal processing. Bomjun J. Kwon (Auditory Professional, Inc., 2801 Logway Rd., Vienna, VA 22181, bjkwon@auditorypro.com)

A scripting language AUX (AUdio syntaX) allows auditory researchers and students with minimal training in programming to generate waveforms and manipulate them with easy and intuitive grammar [Kwon, *Behave. Res.* **44**, 361–373 (2012)]. Akin to the MATLAB, R, or Python environment, AUXLAB is an integrative audio computing environment based on AUX, in which users play and record audio samples, visualize them, write the code for signal analysis and processing, and debug their code. A signal processing task can run on the fly with the microphone, line input to the PC sound card, or streaming audio input. Most notably, AUXLAB is free of charge with open C++ source and does not require special hardware. In this presentation, in addition to introducing newly added AUX language features with object-oriented styles, a hands-on demo of real-time operation emulating a hearing aid's speech processing will be given. Further, the discussion will focus on how this could be used as a low-cost research tool for better hearing aid fitting.

3aPP2. Development and analysis of synoptic evaluation strategy for research platforms for Cochlear Implants and hearing aids. Ram Charan M. Chandra Shekar (Elec. Eng., Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, RamCharan.ChandraShekar@utdallas.edu) and John H. L. Hansen (Elec. Eng., Univ. of Texas at Dallas, Richardson, TX)

Cochlear Implants—Hearing Aids(CI/HA) offer opportunities for researchers wishing to advance algorithm development, since the existing commercial CIs/HAs are closed/sealed for customization, due to intellectual property and safety requirements. In general, Research Platforms (RPs), such as CCi-MOBILE (developed at CRSS-CILab, UTDallas), emulate the functionality of CIs/HAs, and allow researchers to insert their customized sub-algorithms for either benchtop (soundbooth) or field testing. To date, little formalized studies have explored RP performance effectiveness or a formal safety test paradigm. Here, a two phase comprehensive evaluation strategy is proposed that involves acoustical signal processing and user specified electrical stimulation configuration for CCi-MOBILE. While the full evaluation strategy is tedious (+407 h of audio testing: Acoustical test {380 h}, Stimulation parameter test {27 h}), an agile reduced safety test and researcher performance evaluation paradigm is developed by sub-sampling the comprehensive test and system evaluation parameter space. The synoptic safety and performance evaluation paradigm is evaluated to selectively prioritize test conditions based on: (i) acoustical diversity and electrical current level characteristics and (ii) CI stimulation configuration test conditions using physical limits of RF pulses. The proposed synoptic safety and performance evaluation paradigm is shown to be efficient, effective, and ensures reliability of each potentially deployed RP in the field.

3aPP3. Noise management features of the open speech platform. Mingchao Liang (Elec. and Comput. Eng., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, m3liang@eng.ucsd.edu), Kuan-Lin Chen, Wenyu Zhang, Ching-Hua Lee, Bhaskar D. Rao, and Harinath Garudadri (Elec. and Comput. Eng., UC San Diego, La Jolla, CA)

In this contribution, we present the noise management features of the Open Speech Platform (OSP) for hearing aid (HA) research. OSP includes basic HA modules (i) subband decomposition, (ii) wide dynamic range compression and (iii) adaptive feedback cancellation; a baseline single-channel speech enhancement (SE) based on Wiener filtering for noise subtraction. We extended OSP noise management to include a generalized sidelobe cancellation (GSC) beamforming between left and right channels, followed by the SE. Time is the scarcest resource, followed by central processing unit (CPU) resources in commercial HAs. With the proposed GSC+SE approach, we were both time and CPU limited. Release 2019a of OSP has 5.6 ms end-to-end latency and GSC requires an additional 5 ms, putting us above the 10 ms requirement. The wearable device of OSP has 4 cores, C0 – C3. C0 is used for all non-realtime tasks such as kernel, embedded web server, etc. and remaining cores are used for realtime tasks. Naive realization of GSC results in one or more cores not meeting the realtime constraints, resulting in audible artifacts. We present optimizations to meet time and CPU budgets; and preliminary objective and subjective results of the proposed system.

3aPP4. Researcher and user interfaces for studies of hearing-aid self-adjustment. Arthur Boothroyd (Speech Lang. Hearing Sci., San Diego State Univ., 2550 Brant St., San Diego, CA 92101, aboothroyd@cox.net), Carol L. Mackersie (Speech Lang. Hearing Sci., San Diego State Univ., San Diego, CA), Harinath Garudadri, and Tamara Zubaity (Eng., Univ. of California, La Jolla, CA)

The "Goldilocks" explore-and-select protocol of Boothroyd and Mackersie was adapted for field studies of hearing-aid self-fitting and incorporated into the wearable UCSD open-source speech-processing platform (OSP). A researcher interface allows adjustment of hearing aid parameters, including starting response for self-adjustment; compression parameters; and number, consequence, and step size of listener controls. The listener-interface provides for adjustment of level and spectrum using either three controls (overall gain plus low- and high-frequency tilt) or two controls (overall gain and overall spectral tilt). Response selections made in specific acoustic environments can be saved, named, and recalled. Both interfaces are accessible via web-browser on any web-enabled device, including a listener's smart-phone. Every listener-adjustment is logged and options are available for logging information about the ambient noise plus the listener's reasons for, and satisfaction with, response modifications. These interfaces were created using HTML, CSS, and Javascript, and use PHP to communicate information with the embedded webserver on the wearable platform. This web framework is faster and easier to customize than is a native application, and it supports the iterative design required in research. Instructions for customizing the interface are included. They vary in challenge, from editing a text document to modifying documented, readable code.

3aPP5. Open speech platform: Web-apps for hearing aids research. Tamara Zubatiy (Qualcomm Inst. of Calit2, Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, tzubatiy@ucsd.edu), Uposhanto Bhattacharya, Ziqi Gan (Comput. Sci. and Eng., Univ. of California San Diego, La Jolla, CA), Sean Hamilton, Chockalingam Ganz, and Harinath Garudadri (Qualcomm Inst. of Calit2, Univ. of California San Diego, La Jolla, CA)

Open Speech Platform (OSP) for hearing aids (HA) research comprises a realtimemaster hearing aid (RT-MHA); and an embedded web server (EWS) that serves webpages to any browser enabled device for monitoring and controlling RT-MHA. In this contribution, we present 4 classes of web-apps that can be combined and extended in novel ways to conduct psychophysical investigations beyond what is currently possible. (1) The Researcher apps provide access to all RT-MHA parameters; these settings can be saved in named files and recalled easily. (2) HA Fitting apps are written to capture audiologists' hypotheses on HA parameters and their interactions in improving performance; conversely, they can incorporate human-in-the-loop research wherein, the user is forced to select one of two HA parameters (say, A and B) used to process specific speech stimuli stored in a database on OSP; the settings A and B are successively refined based on the user's choice. (3) Ecological Momentary Assessment (EMA) apps are used to capture the user's state for a given HA settings in a given listening environment. (4) Assessment apps enable various tests (e.g., syllable and word recognition tests) in a repeatable manner using stimuli stored in a database on OSP.

3aPP6. Electroacoustic evaluation of pediatric-focused hearing assistive devices/systems in different digital signal transmission arrangements. Tz-Ching Kao (Behavioral and Brain Sci., Univ. of Texas at Dallas, 990 Loop Rd. SW, Apt3.108, Richardson, TX 75080, txk170530@utdallas.edu) and Linda Thibodeau (Behavioral and Brain Sci., Univ. of Texas at Dallas, Richardson, TX)

Pediatric-focused hearing assistive devices/systems (HADS) are intended to facilitate hearing by providing amplification of an acoustic signal and/or improving the signal-to-noise ratio (SNR) for children with hearing loss. Digital modulated (DM) signal transmission systems operate on a 2.4 GHz bandwidth and transmit signals directly from the speaker to the listener. Little is known about changes in electroacoustic performance of the hearing aid when receiving digital transmitted signals. The purpose of this study was to examine the electroacoustic characteristics of pediatric-focused HADS in three digital transmission arrangements with two transmitters and two behind-the-ear hearing aids. The three arrangements included (1) direct audio input (2) proprietary digital streaming, and (3) induction loop. Comparisons were made per ANSI/ASA S3.47 standard for measurement of HADS as well as the electroacoustic transparency based on 2011 American Academy of Audiology Clinical Practice Guidelines. Electroacoustic analysis across HADS in different digital transmission arrangements revealed variable results. Adjustments were necessary for some arrangements for optimal signal transmission. This supports the critical need of electroacoustic evaluation for HADS.

3aPP7. A biologically oriented sound segregation algorithm. Kenny Chou (Biomedical Eng., Boston Univ., 44 Cummington St., RM403, Boston, MA 02215, kfchou@bu.edu), H. Steven Colburn, and Kamal Sen (Biomedical Eng., Boston Univ., Boston, MA)

Listening in an acoustically cluttered scene is a difficult task for both machines and hearing-impaired listeners. Normal-hearing listeners can accomplish this task with relative ease by grouping certain sounds together (i.e., segregating sound sources), then selecting an acoustic target to attend to. An assistive listening device that mimics the biological mechanisms underlying this behavior may provide an effective solution for those with difficulty listening in acoustically cluttered environments (e.g., a cocktail-party). Here, we present a binaural sound segregation algorithm based on the spatial listening mechanisms of the barn-owl. Sounds from the left and right channels are first filtered with an ERB filterbank and then binaural cues are computed, which stimulate model neurons tuned to corresponding frequency channel and spatial locations. Then, the spiking response of specific model neurons are reconstructed into audible waveforms. We evaluate the performance of the proposed algorithm using psychoacoustics, with normal-hearing listeners. Finally, we compare the performance of the proposed algorithm to that of a 16-microphone acoustic beamformer. The proposed algorithm serves as the first step (segregation) in a biologically based solution to the so-called "cocktail party problem."

3aPP8. Evaluation of digital ear scanning for custom hearing protection devices. J. R. W. Stefanson (Warfighter Performance Group, US Army Aeromedical Res. Laboratory/ Hearing Ctr. of Excellence, 6901 Farrel Rd., Fort Rucker, AL 36362, earl.w.stefanson.ctr@mail.mil) and William A. Ahroon (Warfighter Performance Group, US Army Aeromedical Res. Lab., Fort Rucker, AL)

Custom earpieces (e.g., hearing aids, insert hearing protection devices) are traditionally made using physical earmold impressions. Advances in technologies have led to the development of digital methods to capture ear canal geometries for the production of custom earpieces. Three different optical scanning technologies were evaluated and compared to traditional custom earplug fabrication methods. Custom earplugs were made for 20 volunteers from each of the three digital scanning methods, physical earmold impressions, and digital scans of the earmold impressions. A 3-D printing method was also used to produce custom earplugs from the digital scans. Using a within-subjects design, the hearing protection of each custom earplug was evaluated according to standard methodologies described in ANSI/ASA S12.6-2016, Method A. Subjective questionnaires were used for each custom earplug to assess perceived comfort. Results indicate custom earplugs made from traditional physical earmold impressions were significantly higher attenuating but were the least comfortable compared to all other custom earplug fabrication methods. Custom earplugs made from digital ear scanning methods were not significantly different from each other in terms of attenuation or comfort but were significantly better than the physical impression method for comfort.

Session 3aSA

Structural Acoustics and Vibration, Noise, Physical Acoustics, and Engineering Acoustics: Novel Methods For Energy Dissipation in Structures

Jerry H. Ginsberg, Cochair
Retired, 5661 Woodsong Drive, Dunwoody, Georgia 30338

James G. McDaniel, Cochair
Mechanical Engineering, Boston University, 110 Cummington Mall, Boston, Massachusetts, United States

Chair's Introduction—8:30

Invited Papers

8:35

3aSA1. Physical principles governing the damping of structural vibrations by modification of the environment and by addition of appendages. Allan D. Pierce (Cape Code Inst. for Sci. and Eng., PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net)

This paper discusses how vibrations can be damped by the external environment or by addition of external appendages. The topic is an old one and goes back to Stokes (1845) who considered the damping of pendulums by a surrounding viscous fluid. Damping can also occur because of the radiation of sound into the fluid. In more recent years various authors have considered the possibility of attaching something on the surface of the body, where, by some mechanism, vibrational energy is absorbed within in the appendage. Several widely used concepts are reviewed: the dynamic vibration absorber, constrained layer damping, attachment of fuzzy structures, and attachment of piezoelectric strips. A principal (weak damping) approximation is that the vibrations of the environment and the appendage are the same as if the surface vibrations were unaffected. If energy is held in the vicinity of the surface, then there must be some mechanism by which the energy is dissipated in the appendage, and this dissipation should be higher than in the structure proper. The relevant physical principles vary and must be identified if meaningful estimates of damping are to be made. In some cases, generally because of resonance, the energy density in the appendage is considerably higher than in the structure. Given that the governing equations are linear, the damping is invariably frequency dependent. Paper attempts to succinctly summarize the physical principles and give quantitative guidelines for estimating the damping. [Work supported in part by ONR.]

9:00

3aSA2. Friction dampers—Everything old is new again. Jerry H. Ginsberg (Retired, 5661 Woodsong Dr., Dunwoody, GA 30338-2854, j.h.ginsberg@comcast.net)

The title is suggested by a delightful song and dance scene in the film "All That Jazz" (1979). Due to the author's lack of grace, rather than repeating that performance, the presentation will explore a concept suggested by the friction draft gear patented by George Westinghouse in 1888. It and variants are in wide use throughout the North American railroad system. The idea to be examined exploits the fact that the normal forces on the faces of a wedge are greatly magnified, thereby enhancing the friction force associated with the Coulomb friction model. Because of this behavior a simple oscillator whose moving mass is a spring-loaded wedge can absorb much more vibrational energy than other passive techniques. Dynamic analysis of a one-degree-of-freedom model leads to an algebraic expression for the energy dissipated per unit cycle. That formula indicates that critical damping ratios much greater than one are readily obtainable if the springs are precompressed sufficiently. The simplicity of the underlying physical system implies that it is scalable over a large range, from MEMS to seismic structures. A particularly interesting idea is to fabricate a metamaterial consisting of an array of wedge dampers.

9:25

3aSA3. Dynamics of quasi-periodic metastructures. Massimo Ruzzene (Mech. Eng., Univ. of Colorado Boulder, 1111 Eng. Dr., Boulder, CO 80309, massimo.ruzzene@colorado.edu)

This presentation illustrates results of investigations of quasi-periodic assemblies, which uncover unique properties related to vibration localization in one-dimensional and two-dimensional systems. These can be as interesting if not more useful than the interface modes that are found in periodic structures, as the quasi-periodicity framework provides a consistent methodology that leads to vibration confinement in systems that are not ordered, but are described by deterministic property distributions. Moreover, the study of localization in quasiperiodic structure can be supported by the analysis of *localization landscapes*associated with elliptic operators governing the dynamic behavior of the media under consideration, which suggests that localization may be predictable and tunable in a more deterministic way than originally thought. Beam and plate structures with quasiperiodic arrangements of grounding springs and lumped masses are presented as structural components which support a variety of localized modes and that are suitable for the experimental characterization of the dynamic behavior of these configurations. These observations open new avenues and offer new ideas for concepts that may be applied in the design of structures operating in dynamic environments, whose vibrations need to be localized or isolated from specific regions.

9:50

3aSA4. Global effect of locally applied active damping illustrated by laser vibrometry. Vyacheslav Ryaboy (Light & Motion, MKS Instruments, 1791 Deere Ave., Irvine, CA 92606, vyacheslav.ryaboy@newport.com)

Theoretical analysis suggests that decentralized active vibration damping implementing skyhook control (or advanced versions thereof), while applied locally, reduces vibration level over the whole structure by introducing modal damping. That was, however, never directly confirmed experimentally. Laser vibrometry, measuring vibration in multiple points in quick succession and referencing it to the excitation force, provides this opportunity. This paper reports theoretical considerations and experimental results illustrating reduction of total kinetic energy of all-steel optical breadboard by two active dampers. It had been shown (S. Elliott *et al.*) that the effect of collocated decentralized vibration control is limited by two phenomena: instability and "pinning." These issues are analyzed by direct measurement of kinetic energy for a range of active control gains. Optimal gains minimizing kinetic energy with sufficient stability margin are demonstrated and discussed.

10:15–10:30 Break

Contributed Papers

10:30

3aSA5. Novel vibration absorbers for space payload. Halle Green (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC 20064, greenh@cua.edu), Amelia Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC), Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC), Joseph F. Vignola, and Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC)

The vibration transmitted to the payload of a rocket vehicle during liftoff is large in magnitude and broad in frequency. Delicate parts of a satellite like solar panels, lenses, and other equipment carried to perform specific missions can easily be damaged during liftoff. Passive damping systems are currently used to isolate the delicate payload from vibration. However, energy can be dissipated in a narrow frequency band using a Subordinate Oscillator Array (SOA). The SOA is an array of cantilever beams of different size and mass that are attached on a primary resonant structure. The subordinate beams are substantially smaller than the primary structure. This arrangement allows energy transmitted to the primary structure to be dissipated down scale through the vibration of the array of smaller structures. In this study, an SOA is designed to isolate one face of a CubeSat (a cubic satellite of dimensions $10 \times 10 \times 10$ cm) from the broadband vibrations transmitted during the launch of its rocket carrier. Simulation using COMSOL Multiphysics is used to analyze and quantify the effectiveness of this absorber. These numerical results are compared to scanning laser Doppler vibrometry measurements of prototype SOAs affixed to a plate designed to mimic the CubeSat face.

10:45

3aSA6. 1-D elastic immersive wave experimentation in an aluminum beam. Henrik R. Thomsen (Inst. of Geophys., ETH Zürich, Zurich, Switzerland, henrik.thomsen@erdw.ethz.ch), Miguel Moleron, Thomas Haag, Dirk-Jan van Manen, and Johan O. Robertsson (Inst. of Geophys., ETH Zürich, Zürich, Switzerland)

In immersive wave experimentation, a physical experimentation domain is immersed in a larger numerical simulation such that waves in the physical domain drive the numerical simulation and vice-versa. For elastic media, the interaction takes place through sources and sensors at the free surface, where the wavefield is measured and the immersive boundary condition (IBC) is computed and applied. We present a theoretical and experimental study on the implementation of 1-D IBCs for elastic waves in an aluminum beam. Utilizing a 3-D Scanning Laser Doppler Vibrometer (LDV), we measure longitudinal and shear components of a wavefield along the beam. The recorded wavefield is then separated into incident and reflected components and converted into traction. By applying the incident wavefield traction as a boundary condition using a three-component piezo-actuator, we can effectively mitigate broadband longitudinal and shear wave reflections from the boundary, provided the piezo-actuator is calibrated to behave as an idealized point force source. Furthermore, we dynamically link the physical experiment with a numerical background model, introducing virtual scattering. Our results enable elastic immersive wave experimentation at lower frequencies closer to those encountered in real-world scenarios.

11:00

3aSA7. Rotational invariance in the design of structures with imbedded acoustic black hole vibration absorbers. Micah R. Shepherd (Appl. Res. Lab, Penn State Univ., PO Box 30, Mailstop 3220B, State College, PA 16801, mrs30@psu.edu) and Cameron A. McCormick (Appl. Res. Lab, Penn State Univ., State College, PA)

The acoustic black hole (ABH) effect, where the wavespeed of a structure is intentionally reduced to near zero by altering its thickness according to a power-law taper, has been utilized on plates and beams to achieve vibration reduction. Several ABH designs have been proposed for uniform beams including the inverted and the double leaf ABH. However, these designs are only applicable for reducing flexural vibration in a single dimension. A rotationally invariant concept is presented where the imbedded ABH is symmetric around the center line of the beam. Various damping configurations are then applied to the rotationally invariant ABH and their respective amounts of vibration reduction are compared. Finally, the challenges of manufacturing such an ABH design will be discussed in the context of three-dimensional printing and additive manufacturing.

11:15

3aSA8. Parametric study on shape of two-dimensional acoustic black holes. Shuai Cao (Dept. of Mech. Eng., National Univ. of Singapore, 9 Eng. Dr. 1, Singapore 117575, Singapore, caoshuai@u.nus.edu), Kian Meng Lim (Mech. Eng., National Univ. of Singapore, Singapore, Singapore), and Heow Pueh Lee (Mech. Eng., National Univ. of Singapore, Singapore, Singapore)

With the deployment of the passive damping in Acoustic Black Holes (ABH), the effectiveness of ABH in absorbing vibration are improved, as demonstrated in beam-plate structures. Apart from previous studies on the one dimensional ABH (by changing parameters such as the length, power-index of ABH thickness profile, and distribution of damping layer), this paper presents the study of the influence of the shape of a two-dimensional ABH on its absorptive performance. Keeping the area and amount of damping constant over the ABH region, different shapes of ABH embedded in the semi-infinitely long plate were simulated and studied using a finite element model. Dissipated power ratio (DPR) and root-mean-square velocity (RMSV) of the ABH region were measured to evaluate the ABH dissipative behaviors. Structural intensity was also used to visualize and analyze the flow of vibration energy in the system. These analyses of SI propose the correlation between the energy flow and the absorptive performances of ABH. This link between the energy flow and the shape of ABH provides a way to design an ABH for better absorption and dissipation of vibroacoustic energy in the system.

Session 3aSC

Speech Communication: Speech and Hearing Disorders and Child Speech (Poster Session)

Yoonjeong Lee, Chair
University of California, Los Angeles, Head and Neck Surgery, Los Angeles, California 90095

All posters will be on display from 9:00 a.m. to 12:00 noon. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 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

3aSC1. Control modeling toward understanding articulatory disfluency in autism spectrum disorder. Tanya Talkar (Speech and Hearing BioSci. and Technol., Harvard Univ., 244 Wood St., Lexington, MA 02412, ttalkar@g.harvard.edu), Adam Lammert, and Thomas Quatieri (BioEng. Systems and Technologies, MIT Lincoln Lab., Lexington, MA)

Speech control models founded on principles of neuroscience have potential to create powerful clinical tools for personalized medicine and for understanding behavior. We explore two prominent speech control models: Directions into Velocities of Articulators (DIVA) (Guenther, 2006) and Task Dynamics (TD) (Saltzman, 1991) toward an understanding of speech disfluency of autism spectrum disorder (ASD). To introduce perturbations into both systems in the common framework of (Parrrell, 2018), we translate the TD model into MATLAB Simulink. For word stimuli, we compare both models' acoustic outputs, under the hypothesis that speech impediments in ASD arise from overreliance on delayed sensory feedback (Lin, 2015). For DIVA, the output from a 50ms auditory feedback delay contain increased formant production errors and formant oscillations at about 10Hz, sometimes observed in disfluent ASD cases, but also seen in stuttering (Civier, 2010). The same delay introduced in the proprioceptive and tactile feedback in TD takes 5x the time to settle at the target, creating about 20Hz oscillations in TD articulators. Though neither fully capturing disfluency in ASD, together these models lay the groundwork for a more complete neurocomputational speech control model that can be used to longitudinally track and potentially guide ASD speech therapy.

3aSC2. Predicting the impact of hearing aid processing on speech intelligibility. Robyn Hunt (Inst. of Sound and Vib., Univ. of Southampton, B19/1017, University Rd., Southampton, Hampshire SO17 1BJ, United Kingdom, rmh1g13@soton.ac.uk), Steven Bell, and David Simpson (Inst. of Sound and Vib., Univ. of Southampton, Southampton, United Kingdom)

Speech intelligibility is usually assessed using subjective tests, but several objective measures of speech intelligibility have also been proposed. This study compares intelligibility of IEEE sentences in collocated speech-shaped noise, recorded through a low-cost amplifier and three current-issue NHS hearing aids with single-channel noise reduction settings switched on and off. Results from twenty-one normal hearing listeners indicates that single-channel noise reduction algorithms can significantly improve intelligibility of noisy speech ($p < 0.05$), but no differences can be seen between different hearing aid models currently available on the NHS. The low-cost, off-the shelf hearing amplifier performed significantly worse than the NHS prescribed devices. Three objective speech intelligibility measures were applied to the recordings. None of the objective metrics was found to accurately predict the outcomes seen in this study for all hearing aid conditions, though general trends can be predicted ($r = 0.83$ overall, but for any discrete signal-to-noise ratio across all conditions, maximum $r = 0.64$). Machine-learning approaches were used to refit parameters from the best performing of these metrics. Although performance was improved, problems in predicting particular conditions remain, suggesting that objective measures do not encompass all relevant factors. [Action on Hearing Loss (England/Wales Registered Charity Number 207720).]

3aSC3. Effectiveness of a portable device on dysarthrias in daily life—Verification of the effects with DAF and without DAF. Eiji Shimura (Dept. of Medical Sci. Major of Communications Disord. and Sci., Aichi Skukutoku Univ., 2-9 Katahira, Nagakute-shi, Aichi, Nagakute 480-1197, Japan, eshimura@asu.aasa.ac.jp)

Several methods for rehabilitating speech rate control have been widely used to improve speech intelligibility in patients with mild dysarthria. However, few studies have reported the therapeutic effects of these methods in daily life. Delayed auditory feedback (DAF) is one such speech rate control method. A small, portable DAF device has recently been developed that can be used by patients in their daily life. In this study, three dysarthric patients wearing the portable DAF device underwent a 20-min practice per day for 3 months. During the practice, they were instructed to prolong vowel length while using the device. The improvement in the intelligibility of spontaneous conversation was more in patients wearing the portable DAF device than in those who did not wear the device. The patients mastered the speech method using prolonged vowel length pronunciation with the use of the device continuously over a period of time. This finding suggests that the portable DAF devices can be sustainably used in daily life. Even when the device was not used, patients' speech rates significantly decreased at the word level after treatment. This finding suggests that the effectiveness of the portable DAF device is carried over after 3 months of its use.

3aSC4. Duration and amplitude of tone production in children with cochlear implants. Jing Yang (Commun. Sci. and Disord., Univ. of Wisconsin-Milwaukee, 2400 E Hartford Ave., Enderis 873, Milwaukee, WI 53211, jyang888@uwm.edu) and Li Xu (Commun. Sci. and Disord., Ohio Univ., Athens, OH)

The purpose of the present study was two-fold: (1) to examine whether Mandarin-speaking children with CIs showed distinctive durational and amplitude features for the four lexical tones in their tone production; (2) to compare the duration and amplitude patterns of Mandarin lexical tones in monosyllables produced in citation form between CI children and age-matched normal-hearing (NH) children. The participants included 14 prelingually deafened Mandarin-speaking children with CIs and 14 NH children, all aged between 2.9 and 8.3 years old. Each participant produced five CV syllables (fa, fu, pi, xu, ke) in four tones through a tone drill activity. The vowel duration and rms amplitude values at nine equidistant time locations over the vowel duration were obtained. The results revealed that the CI children can produce distinctive duration and amplitude features for the four lexical tones. Their durational pattern and amplitude contours were highly similar to the NH children on tone 1, 2, and 4 but differed from the NH

children on tone 3. In addition, NH children showed positively correlated amplitude contour and F0 contour but the CI children demonstrated inconsistent amplitude and F0 contours for tone 3. This finding suggested that the amplitude contour of a tone and the F0 contour of the same tone may not always be highly correlated.

3aSC5. Distribution of Tourette's verbal tics produced during active speech. Mairym Llorens (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089-1693, llorensm@usc.edu)

Tourette syndrome causes individuals to produce unwanted movements and vocalizations called tics. Verbal tics are those tics that closely mimic words/phrases—their execution requires the same set of articulators needed to produce speech. Tics respond to a preceding urge to tic and are akin to other urge-based actions like cough. Urges to tic can arise at any time but speech and tics must be deployed sequentially as they cannot be produced simultaneously. How are conflicts between the speech and tic motor systems resolved? This study tests the prediction that tics interfere with production of words and intonational phrases less frequently than expected by chance. As predicted, pilot data from one subject showed that the expected probability of words and tics co-occurring were these events independent was significantly higher than the observed likelihood of tics interrupting words. Correspondingly, the likelihood of intonational phrase-final boundaries co-occurring with tics if these events were independent was significantly lower than the observed probability of tics given intonational phrase-final boundaries. Data from multiple people with Tourette's will be presented. Results suggest that utterances are "protected" from tic interruptions, providing evidence that the tic and speech motor systems are linked in production. [Work supported by NIH and NSF.]

3aSC6. The relationship between vowel space area and intelligibility of speech in children with velopharyngeal insufficiency. Hedieh Hashemi Hosseinabad (Commun. Sci. and Disord., Eastern Washington Univ., Spokane, WA), Suzanne E. Boyce (Univ. of Cincinnati, Cincinnati, OH), Ann W. Kummer (Cincinnati Childrens Hospital, Cincinnati Childrens Hospital, Cincinnati, OH, ann.kummer@cchmc.org), Karla Washington (Univ. of Cincinnati, Cincinnati, OH), and Winter Taite (Washington State Univ., Spokane, WA)

Speech of individuals with velopharyngeal insufficiency is characterized by nasalized vowels. It is well known that velopharyngeal coupling significantly alters the acoustic spectrum of vowels. Aside from changes in formant amplitude and bandwidth, formant frequencies may be modified. Centralized F1 and lowered F2 due to effects of nasalization could change the dimensions of the quadrilateral space. Such variations might cause the vowel space area to be compressed which might contribute to speech unintelligibility. In pursuing research on speech intelligibility in cleft palate, this study was conducted to address the relation between speech intelligibility and spectral characteristics of American English vowels in nasal speech. The hypotheses is that narrower vowel space will result in higher degrees of unintelligibility. Ten children aged 4-12 years with different degrees of hypernasality repeated words with four corner vowels. F1 and F2 midpoint frequencies were measured, and vowel space areas were calculated using the formula for an irregular quadrilateral. In order to determine intelligibility, seventy naïve listeners were asked to orthographically transcribe the sentences produced by the participants. Correlational analysis were used to determine the relationship between the vowel space and intelligibility deficits. Further results will be discussed in the meeting.

3aSC7. Individual differences across caregivers in acoustic implementation of infant-directed and adult-directed speech: Modeling impacts on intelligibility in children with cochlear implants. Meisam K. Arjmandi (Dept. of Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd. Oyer Speech & Hearing, Rm. 211A, East Lansing, MI 48824-1220, khalilar@msu.edu), Derek Houston (Dept. of Otolaryngol.–Head and Neck Surgery, The Ohio State Univ., Columbus, OH), Mario Svirsky (Dept. of Otolaryngology-Head and Neck Surgery, New York Univ., New York, NY), Yuanyuan Wang (Dept. of Otolaryngol.–Head and Neck Surgery, The Ohio State Univ., Columbus, OH), Matt Lehet (Dept. of Communicative Sci. and Disord., Michigan State Univ., Pittsburgh, PA), and Laura Dilley (Dept. of Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Caregivers modify their speaking style from adult-directed speech (ADS) to infant-directed speech (IDS) when talking to infants. However, it is unclear how individual caregivers acoustically implement differences between ADS and IDS, and how these differences may affect experienced speech intelligibility by infants, particularly those with cochlear implants (CIs). Seven female talkers spoke fifteen utterances both in IDS and ADS. We analyzed these utterances and their cochlear implant-simulated versions (using 22-channel noise vocoders) to investigate how acoustic distances between ADS and IDS varied across talkers (based on Mel-frequency cepstral coefficients, MFCCs) and how the effect of these shifts in speaking style on utterance intelligibility were different between talkers (using the acoustic index of speech-to-reverberation modulation energy ratio tailored to CI devices, SRMR-CI). Results showed substantial variability across talkers comparing ADS and IDS for caregivers' acoustic profiles from MFCCs and for speech intelligibility from SRMR-CI. These findings suggest that acoustic choices by individual mothers may differentially affect recoverability of intelligible words from speech signals by children with CIs, which may contribute to differences in these children's language outcomes. [Work supported by NIH grant R01DC008581.]

3aSC8. Acoustic variability in electrolaryngeal speech. Steven R. Cox (Commun. Sci. and Disord., Adelphi Univ., Hy Weinberg Ctr. 136, Garden City, NY 11530, scox@adelphi.edu), Christine H. Shadle, and Wei-Rong Chen (Haskins Labs., New Haven, CT)

The electrolarynx (EL) is a hand-held electronic device that provides individuals with a means of communicating verbally postlaryngectomy. The EL produces a vibratory sound source that can be transmitted through the neck, where it excites vocal tract resonances generating speech. While some users attain a high level of intelligibility, the sound is unnatural due to numerous acoustic defects. In this study users of two EL devices were compared: the Servox emits a constant F0; the TruTone's F0 can be varied by the user. The 10 users studied (5 Servox, 5 TruTone) used their EL for at least 24 months and were judged to be proficient. Though a previous study [Watson and Schlauch, *Am. J. Speech Lang. Pathol.*, 18(2) (2009)] showed that intelligibility increased for a user with variable F0, TruTone use was not correlated with greater intelligibility in this study, perhaps because these TruTone users did not vary F0 much (F0 range was 3 to 20 Hz for a corpus of Harvard sentences). Detailed acoustic analyses of EL speech were considered along with other factors such as age and duration of EL experience. Results suggest directions for refinement of EL devices and training protocols for both types of EL users.

3aSC9. Impact of cognitive load and sentence predictability on cognitive spare capacity in elderly adults with hearing loss. Cynthia R. Hunter (Speech-Language-Hearing, Univ. of Kansas, 1000 Sunnyside Ave., University of Kansas, Lawrence, KS 66045, c.hunter@ku.edu), David B. Pisoni (Psychol. and Brain Sci., Indiana Univ., Bloomington, IN), and Larry E. Humes (Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

Listening effort is needed to understand speech that is degraded by hearing loss and/or a noisy environment, and this in turn reduces cognitive spare capacity (CSC), the amount of cognitive resources available for allocation to concurrent tasks. Sentence context is known to boost speech perception accuracy, but how does context affect CSC? Here, we examine the impacts of context and cognitive (memory) load on behavioral measures of CSC. Elderly, hearing-impaired adults listened to noise-masked, spoken sentences

in which sentence-final words were either predictable or unpredictable. Each trial began with visual presentation of a short (low load) or long (high load) sequence of to-be-remembered digits. Accuracy and response times for word recognition and digit recall were both facilitated by sentence predictability, indicating that CSC was greater when sentences were predictable. In addition, response times for both words and digits and accuracy for digits were impaired in the high load condition, reflecting decreased CSC under cognitive load. Participants' baseline cognitive capacity (from a pretest of working memory) generally did not moderate these effects. Results support the idea that predictable sentence contexts can support CSC and thereby improve ease of listening in elderly adults with hearing loss.

3aSC10. Degree of vocal fold adduction affects listener perception of simulated laryngeal vocal tremor. Rosemary A. Lester-Smith (Commun. Sci. and Disord., The Univ. of Texas at Austin, 2504A Whitis Ave. Stop A1100, Austin, TX 78712, rosemary.lester-smith@austin.utexas.edu) and Brad H. Story (Speech, Lang., and Hearing Sci., The Univ. of Arizona, Tucson, AZ)

Laryngeal vocal tremor (VT) is a neurogenic voice disorder characterized by modulation of the fundamental frequency (f_o) and intensity. The primary medical treatment for VT is laryngeal botulinum toxin injections, which result in temporarily reduced speaker- and listener-perceived VT severity. These injections also cause temporary breathiness, which is conventionally considered an adverse effect of the treatment. However, previous studies using a computational model of VT revealed that listeners perceived modulated voices as less "shaky" when the vocal quality was breathy, even when the extent of f_o modulation was the same. The purpose of the current study is to assess the effect of breathiness on listener perception of VT across a range of modulation extents. A kinematic model of the vocal folds and wave-reflection model of the vocal tract were used to simulate VT with degrees of vocal fold adduction representing a spectrum of normal to breathy voice and with f_o modulation extents ranging from 0%–10%. Normal hearing listeners will be presented with pairs of stimuli differing by degree of vocal fold adduction and will be asked to identify which vowel is "shakier." The findings of this study could inform selection of treatment targets and candidates for behavioral therapy for VT.

3aSC11. The development of tone and intonation in Mandarin-speaking children—A pilot study. Jie Yang (Communicative Disord., Texas State Univ., 200 Bobcat Way, Round Rock, TX 78665, j_y90@txstate.edu) and Barbara Davis (Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX)

Mandarin is a tone language in which fundamental frequency (F0) is used to contrast lexical meaning and to convey intonation information. When tone and intonation interact, speakers maintain the shape of the F0 contour for word meaning contrasts but manipulate the magnitude of F0 change to meet intonation requirements. This pilot study provided a new perspective on tone acquisition by exploring phonetic characteristics of tone in different intonation contexts by Mandarin children who have acquired phonological contrasts. Three-, five- and eight-year-old monolingual Mandarin-speaking children and adults participated in this pilot study. Production of monosyllabic Tone 2 (rising F0) and Tone 4 (falling F0) words in isolation and in carrier sentences with interrogative (rising) and declarative (falling) intonation were elicited. Magnitude of F0 change, duration and intensity of the target words were acoustically measured and analyzed. Results suggested (1) a physiological basis of phonatory production where Tone 2 and falling intonation were produced more accurately; (2) lexical first strategy in the youngest children who prioritize tone accuracy over intonation accuracy; and (3) usage of developmentally appropriate and controllable acoustic cues first (e.g., Duration).

3aSC12. Vowel discrimination in noise with formant enhancement: Effects of hearing loss and aging. Jingjing Guan (Texas Tech Univ., Lubbock, TX) and Chang Liu (Commun. Sci. and Disord., The Univ. of Texas at Austin, 1 University Station A1100, Austin, TX 78712, changliu@utexas.edu)

The aim of this study was to investigate whether formant enhancement of F2 would improve vowel formant discrimination in noise for three groups of listeners (i.e., older listeners with hearing loss, older listeners with normal hearing, and young listeners with normal hearing). Thresholds of vowel formant discrimination were examined for the F2 frequencies of three American English vowels: /ʌ/, /ɛ/, and /i/ in long-term speech-shaped noise with signal-to-noise ratio of +6 dB and + 12 dB. F2 frequencies of three vowels were enhanced by 6 dB. Thresholds of vowel formant discrimination were significantly highest for older listeners with hearing loss, while young listeners with normal hearing had lowest thresholds with the older listeners with normal hearing in between. However, discrimination thresholds were improved significantly for all three groups after F2 enhancement was implemented regardless of low or high SNR of background noise. Hearing loss and aging resulted in higher thresholds of vowel formant discrimination in noise and F2 enhancements could improve discrimination performance for all listeners in noise.

3aSC13. Effects of talker variability on spoken word recognition for adult cochlear implant users. Terrin N. Tamati (Otolaryngol., The Ohio State Univ., 915 Olentangy River Rd., Ste. 4000, Columbus, OH 43212, terrintamati@gmail.com) and Aaron C. Moberly (Otolaryngol., The Ohio State Univ., Columbus, OH)

Cochlear implants (CIs) have been highly successful in providing a restored sense of hearing to profoundly deaf individuals. However, CI users must rely on a signal that is highly degraded in acoustic-phonetic details, conveying limited talker information. Normal-hearing (NH) listeners devote processing resources to accommodating talkers' voices, and show less accurate and slower word recognition in multiple-talker compared to single-talker conditions. CI users may display a different strategy in adapting to new talkers due to limitations in talker discrimination. The current study examined the effects of talker variability on spoken word recognition by experienced, post-lingually deafened adult CI users, and the extent to which talker variability effects are mediated by limitations in talker discrimination. Word recognition accuracy for lexically easy and hard words were obtained in single-talker and multiple-talker conditions, varying by gender (female, male, mixed). Results demonstrated that single-talker performance was highly variable, but a detrimental effect of talker variability was observed in mixed conditions, when talker differences were best discriminated, particularly for hard lexical items. Taken together, findings suggest that while CI users may be affected less by talker variability overall, they may display a similar strategy as NH listeners for talker adaptation when talker differences are discriminable.

3aSC14. A comparison between the Korean digits-in-noise test and Korean speech perception-in-noise test in normal-hearing and hearing-impaired listeners. Jae-Hyun Seo (Dept. of Otolaryngology-Head and Neck Surgery, The Catholic Univ. of Korea, 222 Banpo-daero, Seocho-gu, Seoul 06591, South Korea, revivalseo@gmail.com) and Yonghee Oh (Dept. of Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

The purpose of the present study was to determine whether the diagnostic efficacy of Korean version of the DIN (K-DIN) test is equivalent to that of the Korean speech perception-in-noise (K-SPIN) test, which is representative tool for speech-in-noise test. Twenty-seven subjects (15 normal-hearing listeners and 12 hearing-aid users) participated. 50 target digit triplets were presented at the most comfortable levels to each subject while presenting speech-shaped background noise at four levels of SNRs. K-SPIN test also was conducted at same procedure to the K-DIN test. The results of K-DIN and K-SPIN tests were compared by a Pearson-correlation test. The K-DIN test and the K-SPIN test were performed 74 times for each monaural and bilateral listening conditions. The results showed a statistically significant correlation between two tests in all listening conditions (left: $r = 0.788$, $p < 0.001$; right: $r = 0.814$, $p < 0.001$; bilateral: $r = 0.727$, $p < 0.001$). In the time consuming, the K-DIN test and the K-SPIN tests were about 5 minutes and 30 minutes, respectively. The current findings support that the K-DIN test is significantly correlated with the K-SPIN test in hearing-in-noise test performance, which implying that the K-DIN test can be used as a simpler and time-efficient hearing-in-noise test in Korea.

3aSC15. Acoustic correlates of comorbid vand resonance impairment in individuals with amyotrophic lateral sclerosis. Marziye Eshghi (MGH Inst. of Health Professions, 79 13th St., Apt. 1416, Boston, MA 02129, meshghi@mghihp.edu), Kathryn Connaghan, Sarah Gutz (MGH Inst. of Health Professions, Boston, MA), Mohammad Eshghi (Nagoya Univ., Nagoya, Japan), James Berry (Massachusetts General Hospital, Boston, MA), Yana Yunusova (Univ. of Toronto, Toronto, ON, Canada), and Jordan Green (MGH Inst. of Health Professions, Boston, MA)

Assessment of voice and resonance impairment in amyotrophic lateral sclerosis (ALS) may be challenging due to multi-speech subsystem involvement. Although several acoustic measures have been associated with isolated voice and resonance impairment, their efficacy in the presence of comorbid voice-resonance impairment is unclear. The goal of this work is to determine acoustic features that correlate with perceptual judgment of voice and resonance severity in patients with ALS, and identify measures capable of differentiating phonatory, resonance, and co-occurring impairments. Two listeners rated resonance and voice impairment severity of repetitions of "Buy Bobby a puppy" produced by 26 participants with ALS. Samples were stratified based on perceptual ratings: bulbar asymptomatic, predominantly phonatory involvement (i.e., abnormal voice), predominantly resonatory involvement (hypernasality), and mixed (phonatory and resonance involvement). Groups were compared using resonance (one-third octave analysis) and phonatory (cepstral/spectral) measures. The one-third octave analysis differentiated all groups ($p < 0.05$); the cepstral peak prominence differentiated all groups ($p < 0.01$) except asymptomatic versus mixed; and the low/high spectral ratio did not differ between groups. Findings illustrate the challenges of implementing targeted resonance and voice measures in the presence of multi-speech system involvement, though one-third octave analysis is a promising approach to quantifying voice and resonance impairment in ALS.

3aSC16. Assessing head-and-neck cancer patient speech with the vowel dispersion index. Matthew C. Kelley (Linguist, Univ. of AB, University of AB, 4-23 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, mckelley@ualberta.ca) and Daniel Aalto (Commun. Sci. and Disord., Univ. of AB, Edmonton, AB, Canada)

The present study uses a measure of the dispersion of density throughout the vowel space—called the vowel dispersion index—to assess speech patterns in head-and-neck cancer patients. The vowel dispersion index is based on calculating the total variation of the density values in Story and Bunton's (2017) convex hull representation of vowel space density. Overall, the vowel dispersion index quantifies how much change there is throughout the vowel space density. The vowel dispersion index is calculated and analyzed for a sample of 333 recordings of the zoo passage from 107 head-and-neck cancer patients at different stages pre- and post-surgery. Linear mixed-effects regression suggests that the vowel dispersion index is not greatly influenced by the time elapsed since a patient underwent surgery. In contrast, vowel space area is reduced following surgery. These trends suggest that patients retain control of the dispersion of their vowels throughout the vowel space, even after surgery. Their vowel space area does place a constraint on the degree to which they can disperse their vowel tokens, however. These findings are discussed with respect to phonetic theory, principally, Lindblom's (1990) H&H theory.

3aSC17. Rating speech intelligibility using raw-audio as the input to a deep neural-network. Siyu Chen (Linguist, Univ. of AB, Edmonton, AB, Canada), Matthew C. Kelley (Linguist, Univ. of AB, 4-23 Assiniboia Hall, Edmonton, AB T6G 2E7, Canada, mckelley@ualberta.ca), Daniel Aalto (Commun. Sci. and Disord., Univ. of AB, Edmonton, AB, Canada), and Benjamin V. Tucker (Linguist, Univ. of AB, Edmonton, AB, Canada)

Objective measurement of speech intelligibility is a challenge when working with speech-impaired patients. Speech intelligibility scores (the average transcription accuracy across a set of words or sentences by a listener) are a common way of assessing disordered speech. Human-based measurements are less than ideal due to individual differences in listening ability, the time it takes to collect the measures, and other challenges. The present study investigates deep neural networks for fast, automatic, and objective speech intelligibility scoring of head-and-neck cancer patients. We assessed models using the raw acoustic signal as the input to a network with multiple convolutional layers. It is believed that when the raw acoustic signal is used as the input, a convolutional network acts as a filter bank optimized for intelligibility scoring. We report the model accuracy results of repeated training, testing, and comparison of different model structures. Further, we compare the results using a 10-fold cross-validation approach and report the average correlation between the predicted and actual values.

3aSC18. Pitch accent perception and autistic traits for non-native listeners. Grace Kuo (Foreign Lang. and Literatures, National Taiwan Univ., 1 Section 4, Roosevelt Rd., Taipei, Taipei City 106, Taiwan, graciakuo@ntu.edu.tw)

This study examines the role of the *autistic traits* in predicting the prosodic prominence for non-native listeners. The subjects completed the Autism Spectrum Quotient (AQ: Baron-Cohen *et al*., 2001) questionnaire and participated in the Rapid Prosody Transcription Task (Cole *et al*., 2010), where they marked the prominent words and prosodic boundaries of Barack Obama's political speech recordings (475 s in total). The recordings were previously annotated by two trained phoneticians using the ToBI (Tones and Break Indices) conventions for Mainstream American English (Beckman, 1997). Previous studies have shown that native speakers and L2 learners use different acoustic measures as predictors of their perception of prosodic prominence (Pintér *et al*., 2014; You, 2012). Furthermore, Iao *et al*. (2017) found that individuals who were less sociable were less able to discriminate foreign lexical pitch difference. Therefore, this study takes intrinsic individual difference into account (assessed by AQ) to see its effect on the perception of pitch accents in English for non-native listeners. The overall results will also speak to issues regarding the development of prosody acquisition and the role of *attention* in prosodic processing.

3aSC19. Effects of prosody on acquisition of anticipatory coarticulation by Italian-speaking children. Patrizia Bonaventura (Speech-Lang. and Hearing Sci., Hofstra Univ., 183 Locust Ave. #473, West Long Branch, NJ 00764, patrizia.bonaventura@hofstra.edu), Maura Collins (Speech-Lang. and Hearing Sci., Hofstra Univ., Hempstead, NY), and Magda DiRenzo (Istituto di Ortofonologia, Rome, Italy)

This study investigates patterns of acquisition of anticipatory lingual coarticulation by children in Typical Development (TD) and with Speech Sound Disorders (SSD): contradictory data have detected different degrees of coarticulation in children versus adults. This research brings evidence from Italian-speaking children, and analyzes also effects of prosody on development of coarticulation. 2 Italian-speaking SSD and 2 TD, 6-8 years, pronounced disyllabic non-words of different stress structure ([pV'tV] and [pVtV']), upon elicitation by 2 adults. Both CV and V-V anticipatory coarticulation were measured, by (1) locus equations on CV sequences, indicative of regular patterns of stop-consonant interaction with different vowels; (2) trajectories (extent and duration) and slope index (extent/duration) of V-V F2 transitions. Preliminary results based on locus equations scatterplot and regression lines slope coefficients, show similar degree of coarticulation in TD and SSD versus adults in unstressed CV condition, but higher slope values (greater anticipatory coarticulation) in SSD, in stressed CV position. These findings suggest that (a) similar patterns of emergence of anticipatory coarticulation appear in TD versus SSD versus adults, in unstressed condition, with a slightly greater degree of coarticulation in SSD versus TD (b) prosodic factors seem to affect acquisition of coarticulation in SSD.

3aSC20. Novel acoustic measures of coarticulation reveal morphological planning in child speech. Margaret Cychosz (Dept. of Linguist, Univ. of California, Berkeley, 1203 Dwinelle Hall #2650, Berkeley, CA 94720, mcychosz@berkeley.edu) and Keith Johnson (Linguist, Univ. of California, Berkeley, Berkeley, CA)

How do children plan and represent morphologically complex words throughout childhood? We employed coarticulation as a lexical planning metric in speakers of South Bolivian Quechua (SBQ), a highly agglutinating language with over 200 suffixes. 10 adult and 50 child (age 5–10) SBQ speakers completed a real word repetition task. V-C coarticulation was measured in the sequences [ap] and [am] in two contexts: within root morphemes (e.g., papa 'potato') or at morpheme boundaries (e.g., llama-pi 'llama-locative'). Acoustic analysis of coarticulation using formant tracking is complicated in child speech because of children's short vocal tracts and high f0. Following Gerosa *et al.* (in IEEE International Conference on Acoustics Speed and Signal Processing Proceedings (2006), Vol. 1, pp. 393–396), we quantified the degree and time course of coarticulation in Mel-frequency spectral differences. Results show that adults distinguished acoustically between phones more at morpheme boundaries than within morphemes, while children did not distinguish. This conclusion is further evidence that adults decompose words and plan them online. The lack of distinction in child speech, however, suggests that children may be accessing these complex words holistically, even in this highly agglutinating language.

3aSC21. The emergence of conversational turns as a function of parent coaching on language. Naja Ferjan Ramirez (Linguist, Inst. for Learning & Brain Sci., Univ. of Washington, Box 357988, Seattle, WA 98115, naja@u.washington.edu) and Patricia K. Kuhl (Inst. for Learning & Brain Sci., Univ. of Washington, Seattle, WA)

Previous research shows that parents' use of parentese, a nearly universal speaking style distinguished by higher pitch, slower tempo, and exaggerated intonation contours, is associated with advances in children's language learning. We recently showed that when parents are "coached" about the use of parentese when their infants are 6- and 10-months of age, they increase their use of parentese in child-directed speech, which has an immediate and positive effect on child babbling at 14 months. The present longitudinal follow-up study illuminates the mechanisms by which parentese enhanced children's language growth in families who received parent coaching on the use of parentese. We demonstrate that, by 18-months of age, coached parents and their infants ($n = 55$) engaged in significantly more conversational turns compared to uncoached parents and infants ($n = 24$), $p = 0.01$, as measured by first-person LENA recordings in families' homes. Furthermore, infants of coached parents showed enhanced language outcomes at 18 and 24 months, as measured by LENA recordings, the MacArthur-Bates Communicative Developmental Questionnaires, and a vocabulary assessment in the laboratory. We propose that parentese enhances parent-child engagement in communicative turn-taking, thereby creating a positive feedback loop, which could further advance children's language learning.

3aSC22. Large-scale acoustic characterization of mid-low vowels across American, British, and Singaporean children. Yuling Gu (New York Univ., 605 W 42nd St., Apt. 58R, New York, NY 10036, yuling.gu@nyu.edu) and Nancy Chen (Inst. for Infocomm Res., A*STAR, Singapore, Singapore)

This study compares how American, British and Singaporean children differ in their production of mid-low vowels. Read speech was collected from 6–13 years old American children (140 speakers, 43 406 utterances), British children (82 speakers, 32 542 utterances) and Singaporean children (192 speakers, 34 457 utterances), with a balanced gender ratio. All three corpora were designed to be phonetically balanced and formant estimates were extracted from the vowel tokens using the Praat software. Our large-scale acoustic analyses show that for TRAP-BATH split vowels, (1) British and Singaporean children both produce these vowels with higher F1, suggesting a relatively lowered tongue height; (2) These vowels have higher F2 (suggesting a more fronted tongue position) for both American and Singaporean children. One-way ANOVA tests reached statistical significance for all these differences; $F(2,411) = 34.59$, $p < 0.001$ for F1; $F(2,411) = 544.13$, $p < 0.001$ for F2. When comparing /æ/ and /ɛ/ productions, British speakers demonstrate the clearest distinction between the two vowels; Singaporean and American speakers both exhibit a higher and more fronted tongue position for /æ/, causing /æ/ to be more acoustically similar to /ɛ/. One-way ANOVA tests and post hoc Tukey's test demonstrated that these differences are all statistically significant too ($p < 0.001$).

3aSC23. The late acquisition of coarticulatory patterns associated with English liquid articulation. Phil J. Howson (Univ. of Oregon, Sidney Smith Hall, 4th Fl. 100 St. George St., Toronto, M5S 3G3, philh@uoregon.edu) and Melissa Redford (Univ. of Oregon, Eugene, OR)

The late acquisition of liquids is well documented and attributed to the simultaneous anterior and posterior lingual constrictions required. But adult-like segmental production also entails the acquisition of coarticulatory patterns. Liquids are interesting in this regard because they have exceptionally strong effects on their surround, including long-distance anticipatory effects. The current study investigated the development of these effects in 5-year-old, 8-year-old, and adult speech using a previously validated perceptually based measurement technique (Redford *et al.*, 2018). Ten participants per age-group produced minimal pair words with /r/ and /l/ onsets in sentences with several unstressed function words preceding the target liquid. Overall results indicated longer distance anticipatory effects for /r/ than /l/. Two group differences were also identified: anticipatory coarticulation of /l/ was equal to /r/ in 5-year-old speech, and greater than anticipatory coarticulation of /l/ in 8-year-old or adult speech. The results indicate distinct timing patterns for English /r/ and /l/. We suggest that the greater anticipatory coarticulation of /l/ observed in 5-year-olds' speech may indicate its default articulation in a running speech context, and that the development of its adult-like articulation therefore requires learning to inhibit extensive preposturing. [Work funded by NIH grant No. R01HD087452.]

3aSC24. Acoustic characterization of Singaporean children's English with American and British counterparts: A case study on approximants. Yuling Gu (New York Univ., 605 W 42nd St., Apt 58R, New York, NY 10036, yuling.gu@nyu.edu) and Nancy Chen (Inst. for Infocomm Res., A*STAR, Singapore, Singapore, Singapore)

We investigate English pronunciation patterns in Singaporean children in relation to their American and British counterparts by conducting acoustic analysis on /l/ and /ɹ/. A total of 110,405 utterances from American (140 speakers), British (82 speakers) and Singaporean (192 speakers) children aged 6–13 were studied. We find that similar to British children, Singaporean children do not lower F3 as much as American children when producing syllable-final /ɹ/s, suggesting a lack of rhoticity. A one-way ANOVA test demonstrated that these differences are statistically significant across speaker groups: $F(2,411) = 750.9$, $p < 0.001$, and a post-hoc Tukey's HSD test showed that all differences in pairwise comparisons are significant too. Interestingly, similarity between Singaporean and British pronunciation patterns are not observed for syllable-final laterals (dark /l/). We observe that Singaporean children's dark /l/s are produced with characteristics distinct from both those of American and British children. While American and British children demonstrate lowering of F1 and F2 in transitions into dark /l/s, these are not exhibited in Singaporean children's pronunciation. One-way ANOVA tests showed that these differences are significant: $F(2, 1519) = 55.09$, $p < 0.001$ for F1; $F(2, 1519) = 118.7$, $p < 0.001$ for F2, and post-hoc Tukey's tests reached statistical significance too.

3aSC25. Preschool[-aged] children's use of perceptual features to identify spoken languages. Reina Mizrahi (Cognit. Sci., Univ. of California San Diego, 2368 Feather River Rd., Chula Vista, CA 91915, rmizrahi@ucsd.edu) and Sarah Creel (Cognit. Sci., Univ. of California San Diego, La Jolla, CA)

One major question in language development is when and how young language learners identify what language(s) they are hearing. Infants can discriminate languages when at least one is familiar. We have previously demonstrated that regardless of language background, 3- to 5-year-olds accurately associate two languages with two individuals (cartoons), but how

they accomplish this is unclear. Here, we test whether bilingual (n = 32) and monolingual (n = 32) children associate perceptual features versus comprehensibility with individuals. During *learning*, children are familiarized with two cartoon characters: one speaks Spanish sentences; the other, English. At *test*, children see both characters and hear four trial types: English sentences; Spanish sentences; English-like nonsense; Spanish-like nonsense. Sense and nonsense sentences are acoustically matched via bidirectional cross-splicing. On each test trial, the child points to the character who they thought spoke, while eye-movements are being recorded. If children associate language-specific phonetic cues with talkers, they will choose English-speaking characters for English-like nonsense, and Spanish-speaking characters for Spanish-like nonsense. However, English monolinguals may associate characters with (non)comprehensibility. If so, they will choose the Spanish-speaking character for *all* nonsense utterances. Findings will reveal how much children use low-level perceptual features in language recognition, and whether this differs between bilinguals and monolinguals.

3aSC26. Audiovisual enhancement in the perception of children's speech. Jacquelyn Karisny, Gisela Smith, Kristi Oeding, Alexandra Hagen (Univ. of Minnesota, Minneapolis, MN), and Benjamin Munson (Univ. of Minnesota, 115 Shevlin Hall, Minneapolis, MN, munso005@umn.edu)

It is nearly axiomatic that audiovisual (AV) speech is more intelligible than audio-only (A-only) speech, particularly when the speech is presented in a challenging listening environment, such as in background noise [e.g., MacLeod and Summerfield, *Br. J. Audiol.*, 2 (1987)]. No previous research on audiovisual speech perception has examined the perception of children's speech. Children may elicit a smaller AV benefit than adults, as their visual articulatory movements are more variable than adults' [e.g., Smith and Goffman, *J. Speech Lang. Hear. Res.* **41** (1998)], and hence are less informative perceptual cues. Alternatively, the overall lower intelligibility of children's A-only speech might lead them to elicit overall higher AV benefits than adults. To examine this question, we collected developmentally appropriate sentence productions from five, 4-6 year old children, and five sex-matched adults. Ongoing work is examining the intelligibility of these sentences in multitalker babble in A-only and AV conditions in a variety of signal-to-noise ratios, so that we can compare AV benefits for children and adults when A-only intelligibility is matched. Both sentence intelligibility and eye gaze during perception are being measured. Results will help us understand the role of individual-speaker variation on the magnitude of AV benefit.

3aSC27. Developmental changes in categorical perception of Mandarin Chinese tones. Dandan Qin (Xian Jiaotong Univ., Xian Jiaotong University, Xian, Shaanxi 710049, China, carissa0304@stu.xjtu.edu.cn), Bing Cheng, Wen Zhang (Xian Jiaotong Univ., Xian, Shaanxi, China), and Yang Zhang (Univ. of Minnesota, Minneapolis, MN)

This study examined developmental changes in categorical perception of Mandarin Chinese lexical tones by 22 adults and 16 10-year-olds, who were all native Mandarin Chinese speakers. The speech stimuli included three types of continuum, tone 1 (T1)/tone 2 (T2), tone 1/tone 4 (T4) and tone 2/tone 4. The experimental protocol used conventional identification and discrimination tasks. The results confirmed categorical perception of the tonal continua by both adult and child groups. Refined tests further revealed the existence of T1 category with two phonetic boundaries in the T2/T4 continuum. More importantly, age-dependent differences were observed in discrimination accuracy and sensitivity to pitch height. Collectively, the data suggest that although adult-like categorical perception of lexical tones can be found by 10 years of age and earlier as the literature suggested, the dynamic developmental trajectory in fine-tuning the categorical perception of lexical tones extends beyond the age of 10.

3aSC28. Temporal extent of anticipatory lip rounding in child and adult speech. Melissa Redford (Linguist Dept., Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, redford@uoregon.edu), Jeffrey Kallay, and Jill Potratz (Univ. of Oregon, Eugene, OR)

Look-ahead models predict that anticipatory lip rounding is defined segmentally: for example, extending backwards from the target to the first segment specified for rounding. Gestural models predict that it is limited by the duration of the activation interval, which is why coarticulatory effects are weaker in slow speech compared to fast speech. This study investigated the temporal extent of anticipatory rounding in child versus adult speech with these model predictions in mind. Child speech is slower than adult speech. It is also purportedly equally coarticulated as adult speech, but long-distance effects are underexamined. Two groups of school-aged children, 5- and 8-year-olds, and one group of adults produced simple subject-verb-object sentences. The vowel of the object noun contained either the same unrounded vowel as the verb or a different rounded vowel. An unstressed determiner intervened between the verb and noun. Coarticulatory extent was measured perceptually using gated AV speech. Rounding was found to extend backwards to the preceding stressed vowel in both child and adult speech, and so up to (roughly) 400 ms before the target in 5-year-olds' speech compared to 340 ms in 8-year-olds' speech and 235 ms in adults' speech. These results are more easily reconciled with look-ahead model predictions than with gestural model predictions. [Work supported by NICHD grant #R01HD087452.]

3aSC29. Acoustic correlates of talker height in children's voices. Abbey L. Thomas (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, Richardson, TX 75080, thomasabbey8@gmail.com), Santiago Barreda (Linguist, Univ. of California, Davis, Davis, CA), and Peter F. Assmann (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, Richardson, TX)

In a recent study, we investigated listeners' ability to judge talker height from CVC syllables spoken by children. The present study extends this research by examining acoustic variables that predict talker height. We analyzed recordings of the sustained vowel /a/ produced by children 5 through 18 years of age, taken from a larger children's speech database. In addition to measures previously found to predict speaker height (GMFF and G0, the geometric mean of the lowest three formant frequencies and fundamental frequency, respectively) we investigated measures related to the glottal source reflecting breathiness and spectral tilt. Multiple regression models incorporating these measures provided accurate estimates of talker height (mean average error of 9.3 cm). Consistent with our previous findings for age, a model specifying the sex of the talker predicts talker height significantly more accurately than a model that excludes sex.

3aSC30. Are there any differences between the tongue and posterior pharyngeal wall movement patterns during normal and Masako swallow?—An ultrasound analysis. Emily Wang (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 600 South Paulina, Ste. 1017 AAC, Chicago, IL 60612, emily_wang@rush.edu), Adam Maxwell (Urology, Univ. of Washington, Seattle, WA), and Leonard A. Verhagen Metman (Neurological Sci., Rush Univ. Medical Ctr., Chicago, IL)

Parkinson's disease (PD) is a neurodegenerative movement disorder affecting volitional movement due to Dopamine-producing neuron death in the substantia nigra. Impaired swallow function is highly prevalent with nearly 100% involvement in PD during the disease course and involves all three phases of swallowing. The oral stage of swallowing is impaired first. Reduced bolus control and reduced interior-to-posterior movement of the bolus transfer lead to delayed swallowing response, premature spillage, decreased base of tongue (BOT) retraction and strength, decreased hyolarygneal elevation and excursion, and reduced action of pharyngeal constrictors. Together, these result in bolus retention, residue throughout the pharynx, and reduced airway protection leading to significant aspiration. To target these deficits we have designed an exercise regime to teach early to mid-stage PD patients to do the Masako swallow maneuver of saliva (swallow with tongue-tip held between teeth), with high intensity and frequency (i.e., 120 repetitions per day) in one month. The preliminary results are very promising. All participant demonstrated positive changes. To understand the underlying mechanism, we used Ultrasound and recorded 20 normal subjects while they did three normal swallows and 3 Masako swallow maneuver. The initial observations indicated the tongue and pharyngeal movement patterns differ between the two maneuvers.

Session 3aSP

Signal Processing in Acoustics and Underwater Acoustics: Memorial Session in Honor of Ed Sullivan

Ning Xiang, Cochair
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Mathematical Sciences, New Jersey Institute of Technology, 323 ML King Blvd, Newark, New Jersey 07102

Chair's Introduction—8:00

Invited Papers

8:05

3aSP1. Synthetic aperture towed-array processing: The Edmund J Sullivan Legacy. James V. Candy (Eng., Lawrence Livermore National Security, PO Box 808, L-151, Livermore, CA 94551, tsoftware@aol.com)

Dr. Edmund J. Sullivan developed synthetic aperture towed-array processing. Confronted by many nay-sayers Dr. Sullivan persisted with his work and developed the first approach as the signal processing group leader (SPGL) at the SACLANT ASW Center (now CMRE) in La Spezia, Italy. The first notional passive synthetic aperture processor using an overlapped correlation method was jointly developed ["Extended Towed Array Processing by Overlapped Correlator," (*JASA*, 1989)]. This idea was to blossom even further in his collaborative works (Stergiopolous *et al.*) extending these ideas to a fully, recursive (in-time) model-based passive-synthetic aperture processor ["Space-time array processing: A model-based approach" (*JASA*, 1997)]. He also began collaborations with the Swedish Navy where he performed joint experiments in model-based passive synthetic aperture evaluating its performance in the ocean and demonstrating its effectiveness. Dr. Sullivan began mentoring more researchers and teaching at the University of Rhode Island where he advised students in underwater acoustics and processing leading researchers to the model-based approach (Cousins), synthetic aperture processing (Edelson) ["On the performance of the overlap-correlator synthetic aperture technique" (*JASA*, 1991)] and broadband processing (Holmes) ["Broadband passive synthetic aperture" (*JASA*, 2006)]. His ideas were summarized in his recent text, [*Model-Based Processing for Underwater Acoustic Arrays* (Springer, 2015)].

8:35

3aSP2. Ed Sullivan and model-based ocean acoustic signal processing. Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd, Newark, NJ 07102, michalop@njit.edu)

Ed Sullivan was a pioneer in model-based signal processing and his contributions have inspired numerous scientists in the fields of acoustic signal processing and ocean acoustics. Work by Ed and his colleagues is among the first efforts to formulate ocean acoustic inverse problems in a sequential filtering framework; methods were developed that overcome the destructive role of mismatch between assumed and true physics in inverse problems. In paper "Model-based ocean acoustic signal processing," Acoustics Today, July 2011, Ed Sullivan identifies three main directions in signal processing, detection, classification, and estimation; he focuses on estimation and presents a study of model-based methods for inverse problems and how those methods can be successfully applied to the field of ocean acoustics. This presentation reviews these applications and discusses the impact of Ed Sullivan's contributions on other advances in ocean acoustic signal processing.

8:55

3aSP3. Sullivan's contributions to model-based acoustic array processing for underwater and beyond. Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu)

Edmund Sullivan's long and fruitful research efforts have spanned a number of fields, both underwater and ashore. Most notable to this author is his achievement on model-based processing for underwater acoustic arrays, documented in a 2015 Springer monograph. Through his unremitting dedication and long-time service to the Acoustical Society of America, and most prominently, the Technical Committee on Signal Processing in Acoustics, Ed's achievements have long garnered utmost respect and reputability. This paper reviews his recent effort on model-based Bayesian signal processing with a rigorous handling of the Kalman filter, and acoustic array processing within the Bayesian framework. This paper also reflects personal remembrance of the mutually beneficial collaboration with Ed Sullivan in acoustic landmine detection and Bayesian analysis.

9:15

3aSP4. Real time 3-D ultrasound diagnostic imaging system including 3-D adaptive beamforming processing. Stergios Stergiopoulos (Elec. and Comput. Eng., Univ. of Toronto, 157 Burnett Ave., North York, ON M2N 1V6, Canada, stergios@stylari.com)

This paper describes the analog front end module and the computing architecture components of a fully digital real-time 3-D ultrasound system. The computing architecture is designed to allow for an efficient implementation of a 3-D adaptive beamformer that has the capability to improve the angular image resolution of a planar array by approximately four times. The complex 3-D beamforming structure is decomposed into two steps of line array beamformers and this kind of decomposition process for the 3-D beamformer allows for its efficient implementation into the highly parallelized multi-processor based computing architecture for real time 3-D ultrasound imaging applications providing 20 volumes per second at a full opening angle of 80 deg (azimuth and elevation). The main objective of this paper is to describe the details of the system processing requirements and design consideration for the computing architecture and provide the experimental results showing that the proposed implementation can achieve the targeted frame rate. An easy to use user interface in combination with a decision-support process provides the possibility for a rapid and automated diagnosis of internal injuries like bleeding or facilitates image guided surgery.

9:35

3aSP5. Multi-frequency sparse Bayesian learning with noise models. Kay L. Gemba (MPL/SIO, UCSD, University of California, San Diego, 8820 Shellback Way, Spiess Hall, Rm. 446, La Jolla, CA 92037, gemba@ucsd.edu), Santosh Nannuru (Signal Processing and Communications Res. Ctr., IIIT, San Diego, CA), and Peter Gerstoft (MPL/SIO, UCSD, La Jolla, CA)

Ed Sullivan's legacy includes significant contributions to the field of signal processing. Inspired by his Bayesian approach, we present results for a method coined Sparse Bayesian learning (SBL) to estimate source parameters. Previously, SBL has been applied to the matched field processing application [K. L. Gemba, S. Nannuru, and P. Gerstoft, "Robust ocean acoustic localization with sparse Bayesian learning," *IEEE J. Sel. Top. Signal Process.* **13**(1), 49–60 (2019)]. This multi-source scenario required adaptive and robust processing, and included a non-stationary noise model. The adaptive SBL algorithm models the complex source amplitudes as random quantities, providing a degree of robustness to amplitude and phase errors. Further, its formulation is flexible and can accommodate advanced noise models. We consider the application of different noise models in simulations and experimental data and compare SBL performance to traditional processing.

9:55–10:10 Break

10:10

3aSP6. Ed Sullivan: Mentor, scientist, colleague, and friend. Geoffrey S. Edelson (Maritime Systems & Technol., BAE Systems FAST Labs, MER15-2350, P.O. Box 868, Nashua, NH 03061-0868, geoffrey.s.edelson@baesystems.com)

I had the great fortune of meeting Ed Sullivan in 1989 when he was the Manager of Basic Research at the Naval Undersea Warfare Center and, to his dismay, was spending almost all of his time managing rather than researching. Ed hired me as a part-time student associate to analyze and implement passive synthetic aperture sonar, the core of our collaborative efforts for over twenty years and the technical focus of this presentation. Conversations with Ed, whether technical or not, were intellectually stimulating, fun, or mostly a combination of the two. Importantly though and through Ed's confident and unique lens, he provided me – and many of you – with insights into highly relevant areas such as matched field and model based signal processing, music, history, and physics!

10:30

3aSP7. A few comments on passive synthetic aperture sonar in honor of Ed Sullivan. Gerald L. D'Spain (Marine Physical Lab, Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu)

I came to know Ed Sullivan after submitting a paper on passive synthetic aperture sonar for publication in the Journal of the Acoustical Society of America. At that time, he was an Associate Editor of the journal. I knew of him, by reputation, of course. Only after the paper reviews came back did I get to know him, the person, the hard-core, old-school signal processor and physicist. At first, I simply tried to humor him to get the paper accepted for publication, figuring that quotes from a few of his papers sprinkled here and there would suffice. But after spending time expanding my knowledge of his work, I developed a deep appreciation for the foundation in signal processing he and colleagues had built. The published version of our paper was significantly improved as a result. This talk presents some of Ed Sullivan's results on model-based signal processing and its application to passive synthetic aperture sonar, along with some of our published results. Its main purpose, however, is to serve as a reminder that we all stand on the foundation built by those who have gone before. [Work supported by the Office of Naval Research.]

10:50

3aSP8. Defense applications of Edmund J. Sullivan's acoustic signal processing concepts and methods. Brian G. Ferguson (Acoust. Systems, Defence Sci. and Technol., Locked Bag 7005, Liverpool, New South Wales 1871, Australia, Brian.Ferguson@defence.gov.au)

Passive acoustic signal processing for defense involves the detection, classification, localization and tracking of sound sources. Ed Sullivan posited that "acoustic signals carry information in their kinematics, as well as their spatial and temporal phase structure, so that including information in the signal model will provide enhanced performance." This idea is applied to the practical defense problem of locating the point-of-origin of hostile sniper fire by processing the acoustic signals sensed by a receiving array. Superior acoustic source localization performance is achieved by including the deceleration of the bullet in the signal model. Next, Ed's passive synthetic aperture methods are applied to the inverse problem of estimating the source (or rest) frequencies and flight motion parameters of turboprop aircraft and helicopters using a single acoustic sensor and then, an array of sensors. In this case, the source signal is temporally coherent and the source kinematics is contained in the received Doppler-shifted signals. Finally, source localization of broadband signals using cross-correlation methods is observed to fail when the different phase structures of the received signals lead to them being uncorrelated, which can be remedied by differential Doppler compensation.

Contributed Papers

11:10

3aSP9. Array signal processing methods for achieving super-resolution using vortex waves. Andrew Young (Acoust. Div., US Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, ayoung.ece@duke.edu), Matthew D. Guild, and Jeffrey S. Rogers (Acoust. Div., US Naval Res. Lab., Washington, DC)

Helicoidal (vortex) acoustic waves have received much interest lately and can be generated by a variety of means, such as applying a prescribed phase shift to the elements of conventional 2-D planar arrays or by employing a leaky wave antenna in a circular arrangement. In the context of array signal processing, the beampattern associated with vortex wave generation has a strong central null that can be exploited to achieve far field super-resolution in a variety of applications. Although recent physical acoustics research into vortex waves has focused on active applications such as particle manipulation or propagating sub-diffraction-limit features into the far field, vortex waves can also be utilized in passive applications such as source localization. This work investigates the signal processing implications of using vortex waves in both active and passive regimes. Key tradeoffs between conventional, adaptive, and vortex-wave-based array signal processing methods are examined by simulating a variety of array geometries and source/target configurations. Results indicate the potential for vortex wave methods to provide performance improvements over conventional and adaptive methods such as MVDR (minimum variance distortionless response) and MUSIC (multiple signal classification) in snapshot-deficient and high-bearing-rate target scenarios.

11:25

3aSP10. Deconvolution of underwater acoustic marine mammal signals from the Northern Gulf of Mexico. Kendal Leftwich (Phys., Univ. of New Orleans, 1021 Sci. Bldg., New Orleans, LA 70148, kmleftwi@uno.edu), Juliette W. Ioup, and Michael J. Haas (Phys., Univ. of New Orleans, New Orleans, LA)

Extracting useful information from electronically recorded data measurements often requires the signal to be separated from the noise and the blurring impulse response deconvolved out. In order to denoise LADC-GEMM underwater acoustic data, we apply and compare several techniques from Fourier frequency filtering and wavelet denoising. We also look at methods of estimating the acoustic impulse response (system response, transfer function, etc.) for the northern Gulf of Mexico to be used in implementing several deconvolution methods applied to the same dataset. Deconvolution techniques tested include Fourier deconvolution, Wiener filtering, various blind deconvolution methods, as well as combination algorithms. Comparisons of techniques will be presented, showing that the different methods may yield different results. Metrics for analysis will include both the final quality of results along with computational efficiency. Some recommendations will be made within the framework of marine mammal signal detection; although specifics of denoising and deconvolution will often be data dependent, these ideas can be applied more broadly to a variety of situations.

11:40–12:00
Panel Discussion

Session 3aUWa

Underwater Acoustics: Sound Interaction with the Seabed

Dajun Tang, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, Washington 98105

Eric I. Thorsos, Cochair
Applied Physics Laboratory, University of Washington, 1013 NE 40th Street, Seattle, Washington 98105

Contributed Papers

8:00

3aUWa1. Reconstruction of volume scattering coefficient with considering seabottom scattering layer. Polina Vornovskikh (Comput. Sci., Mathematical and Comput. Modeling, Far Eastern Federal Univ. (FEFU), 8 Sukhanova St., Vladivostok 690090, Russian Federation, vornovskikh.polina@gmail.com), Andrei Sushchenko (Comput. Sci., mathematical and Comput. modeling, Far Eastern Federal Univ. (FEFU), Vladivostok, Primorskii krai, Russian Federation), and Igor Prokhorov (Inst. of Appl. Mathematics FEB RAS, Vladivostok Primorskii krai, Russian Federation)

A problem of remote sensing in the ocean by a point isotropic sound source is considered. To describe the process of remote sensing of the ocean, the authors selected a kinetic model based on the integro-differential radiation transfer equation. The inverse problem is formulated, which consists in reconstructing the coefficient of volume scattering in the seabottom layer. An equation for simulating the single scattered signal is obtained. Computational experiments are carried out to analyze the solution of the inverse problem.

8:15

3aUWa2. Integral equations for scattering from a rough ocean bottom. Dajun Tang (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

The motivation for this work is to develop a method in order to assess the impact of bottom roughness on transmission loss and reverberation in the mid-frequency range (1–10 kHz). A set of integral equations of the second kind is developed which yields formally exact solutions for sound scattering by a one-dimensional rough interface separating two homogeneous fluid half- spaces. The second kind equations have better convergence properties as compared to the first kind equations and are better suited for iteration methods, which is needed in dealing with scattering from very long rough interfaces at small grazing angles. The current work optimizes the equations to avoid the issue of hyper-singularity. Numerical examples are shown to demonstrate the validity of the method. The well-established but less well known forward- and backward iteration schemed developed by Kapp and Brown for sea surface scattering is extended to the seafloor scenario. It is found that the iteration method is highly effective, and one, or at most two iterations, are enough to yield accurate results for cases involving typical roughness encountered in a natural seafloor. This paves the way for modeling seafloor scattering over large distances and waveguides. [Work supported by ONR Ocean Acoustics.]

8:30

3aUWa3. Trans-dimensional range-dependent geoacoustic inversion using modal dispersion data in the South China Sea. Jin-Bao Weng (Third Inst. of Oceanogr., Ministry of Natural Resource, No. 178 Daxue Rd., Siming District, Xiamen, Fujian 361005, China, wengjinbao@tio.org.cn), Stan E. Dosso, N. Ross Chapman (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Yan-ming Yang (Third Inst. of Oceanogr., Ministry of Natural Resource, Xiamen, Fujian, China), ZhaoHui Peng, GuangXu Wang, and Lingshan Zhang (State Key Lab. of Acoust., Inst. of Acoust., Beijing, China)

This paper presents geoacoustic inversion of modal dispersion data in the South China Sea using a single hydrophone and multiple impulsive sources at ranges from 5–100 km along a shallow-water track with slowly varying bathymetry. As a first step, the source waveform and bubble pulse are deconvolved from the recorded time series using a short-range source recording, corrected for the surface reflection. A time-frequency warping analysis is used to filter individual modes and obtain dispersion (arrival time as a function of frequency) data for three modes. Trans-dimensional Bayesian inversion is applied to the modal dispersion data, based on probabilistic sampling over an unknown number of seabed layers. Range-dependent inversion is considered, based on separating the environment into a sequence of range-independent sections, with frequency-dependent modal propagation times summed over segments. The inversion results are compared to core samples collected at sites along the survey line and to an independent headwave arrival-time analysis of the impulsive-source data.

8:45

3aUWa4. Variability of Doppler shift in the ocean bottom reverberation data from the East Sea of Korea. Young Cheol Jung (Naval Architecture and Ocean Eng., Seoul National Univ., 1, Gwanak-ro, Gwanak-gu, Seoul 151 -744, South Korea, dicaffri@snu.ac.kr), Seongil Kim (Agency for Defense Development, Korea, Changwon, Kyung-nam, South Korea), Keunhwa Lee (Sejong Univ., Korea, Seoul, South Korea), and Woojae Seong (Naval Architecture and Ocean Eng., Seoul National Univ., Korea, Seoul, South Korea)

Sea trials measuring the ocean bottom reverberation using an active towed array sensor (ATAS) equipped with triplet receiver array were conducted in the East Sea of Korea during August, 2015. The continuous-wave waveforms with center frequency 3 kHz and pulse lengths of 0.1, 0.3, and 1 s are used as source signals, respectively. Ship velocity was between 4 and 5 knots and the water depth varies from 500 to 1250 m. Also, sound speed profiles were obtained by expendable bathythermographs (XBTs) during the ship maneuvering, where the depths of minimum sound speed appeared between 208 and 306 m. From the measurement results, we observed variability and spreading of reverberation Doppler shifts, which depends on the bottom bathymetry, platform motion, source depth, and receiver array depth. In particular, significant asymmetry of the Doppler spectrum occurred at the depths of the source and receiver below 120 m. The instantaneous frequency

Doppler of reverberation signal in each beam direction provides information on the bathymetry and scattering characteristics of the bottom. This frequency shift is traced in time and analyzed.

9:00

3aUWa5. On the sound speed dispersion and the frequency dependence of sound attenuation in a fine-grained sediment in the New England Mud Patch. Lin Wan (College of Eng., Univ. of Delaware, 104 Robinson Hall, Newark, DE 19716, wan@udel.edu), Mohsen Badiey (College of Eng., Univ. of Delaware, Newark, DE), David 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)

The study on the sound speed dispersion and the frequency dependence of sound attenuation in marine sediments is of great importance in understanding the physics of sound propagation in the sea bottom. Modal dispersion curves (MDC) have only been utilized to estimate the low-frequency sound speed. At higher frequencies (i.e., above a few hundred hertz), arrival time difference for different modes generally become smaller and MDC is difficult to extract because of the cross-mode interference. This paper applies time-frequency representations (TFR) with high time-frequency resolution (e.g., optimal-kernel and center-affine-filtered TFR) to broadband acoustic signals (e.g., 31g explosive and combustive source signals) detonated along the 15-km Northwest-Southeast tracks during the Seabed Characterization Experiment in the New England Mud Patch, which has a surface fine-grained sediment layer overlying a sandy bottom. The resulting high-resolution broadband (150–1500 Hz) MDC, in conjunction with TL data, are fed to a previously developed two-step dimension-reduced geoacoustic inversion algorithm including modal dispersion-based inversion for sound speed and energy-based inversion for attenuation [*IEEE J. Ocean. Eng.*, **44**, in print, (2019)]. The estimated sound speed and attenuation are compared with the theoretical results from seabed geoacoustic models for fine-grained sediments. [Work supported by ONR Ocean Acoustics (322OA).]

9:15

3aUWa6. Geoacoustic inversion using vector acoustic modal dispersion. Julien Bonnel (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu), David R. Dall'Osto, and Peter H. Dahl (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Geoacoustic inversion is traditionally performed using sound pressure data. Here, we present a study of geoacoustic inversion using particle velocity, as recorded on a vector sensor. The scope of the study is restricted to shallow water and low-frequency impulsive sources. In this context, the propagation is described by normal mode theory. The modal description of the vector field is well known: the mode amplitudes are different than from the pressure field, but their phase is the same. As a result, modes can be filtered from particle velocity data in the same way that they are filtered from pressure data, as long as one considers filtering methods that are based on the modal phase. We show that warping (a non-linear resampling scheme) enables modal filtering on both pressure and particle motion data. Then, the filtered modes (pressure and velocity) are coherently combined to form four vector acoustic metrics [Dahl and Dall'Osto, *IEEE J. Oceanic Eng.* 2019]. Each metric is used as an input for Bayesian geoacoustic inversion. The whole method is illustrated on a simulated scenario that reproduces the Seabed Characterization Experiment (SBCEX) that took place on the New England Mud Patch in 2017. [Work supported by the Office of Naval Research.]

9:30

3aUWa7. Passive acoustic inversion with virtual head waves. Jie Li (Collaborative Innovation Ctr. for Adv. Ship and Deep-Sea Exploration, State Key Lab. of Ocean Eng., Shanghai Jiao Tong Univ., 800 Dongchuan Rd., Minhang District, Shanghai 200240, China, jie_li@sjtu.edu.cn), Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR), and Jun Fan (Collaborative Innovation Ctr. for Adv. Ship and Deep-Sea Exploration, State Key Lab. of Ocean Eng., Shanghai Jiao Tong Univ., Shanghai, China)

A passive acoustic inversion method with virtual head waves extracted from ocean surface noise is presented. The noise is assumed to be coming from the sea-surface due to wind and waves and a vertical array is used to produce beamformed time-series. Cross-correlating a beam steered downward with one steered upward at the same angle produces virtual head waves with periodic peaks in the time-series but only for the beams steered at the seabed critical angle. The virtual head wave has the same phase speed as the real head wave, but travel time is offset, thus virtual. It has been attributed to the cross-correlation between head waves and head waves or head waves and perfect reflections. With two of the virtual head wave travel times and the critical angle, an inverse method is developed to extract the water depth, water column and seabed sound speeds as well as the array depth. The theory is developed for the expected travel times and this provides the basis for the inversion methodology. Both the theoretical framework and the inversion method will be presented along with illustrating examples.

9:45

3aUWa8. Characterization of acoustic attenuation in a coral reef environment. Kayla Thilges (Ocean Eng., Univ. of Rhode Island, 18 Meadowrue Trail, Saunderstown, RI 02874, kayla_thilges@uri.edu), Lora J. Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Simon Freeman (Naval Undersea Warfare Ctr., Newport, RI), and Lauren Freeman (Naval Undersea Warfare Ctr., Washington, DC)

Coral reefs are complicated and understudied acoustic propagation environments. In addition to geometric spreading, there are propagation losses due to bottom attenuation, volumetric scattering, and boundary scattering from bathymetric variability and sea surface roughness. On a Hawaiian coral reef, a field test was performed in shallow water during which low-level tones were projected from an underwater speaker and received by a single hydrophone at various ranges up to 500 m from the source. Acoustic data gathered during the field test were analyzed to characterize the sound propagation environment. Conventional geometric spreading assumptions were challenged for the sloping bathymetry characteristic of the Hawaiian coral reef environment. Geoacoustic parameters of the coral reef environment were extracted from transmission loss measurements using a nonlinear least-squares inversion. Losses were estimated for frequencies of 500 Hz, 2 kHz, 5 kHz, 10 kHz, and 15 kHz.

10:00

3aUWa9. A convolutional neural network applied to measured time series for source range and ocean seabed classification. David F. Van Komen (Phys. and Astronomy, Brigham Young Univ., N283 Eyring Sci. Ctr., Provo, UT 84602, david.vankomen@gmail.com), Tracianne B. Neilsen (Phys. and Astronomy, Brigham Young Univ., Provo, UT), David P. Knobles (KSA, LLC., Austin, TX), Peter H. Dahl, and David R. Dall'Osto (Univ. of Washington, Seattle, WA)

Source localization and environmental inference are common problems in ocean acoustics requiring computationally intensive algorithms and knowledge of the search space. Convolutional neural networks (CNNs) learn useful features for making predictions directly from a gridded input signal circumventing the costly practice of selecting features or comparisons to a forward propagation model. To take advantage of these benefits, a CNN was trained and validated on simulated pressure signals generated using four different environments (sandy, muddy, and mixed-sediment layers) for several ranges (2 to 12 km) to make environment class and range predictions. The trained network was then tested on recorded signals from the APL-UW Intensity Vector Autonomous Receiver system during SBC2017. We found that the network could make predictions with 99% accuracy on the simulated validation dataset but struggled to make unbiased predictions on the measured data. The struggles highlight the need to incorporate all significant features in the training data, such as time alignment and addition of noise. The CNN has the power to make predictions on this type and clearly see the differences in the data, but investigations for creating the proper training dataset or building a deeper network are required. [Work supported from Office of Naval Research.]

Session 3aUWb

Underwater Acoustics and Acoustical Oceanography: A. B. Wood Medal and Prize Lecture

D. Benjamin Reeder, Chair
Oceanography, Naval Postgraduate School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, California 93943

Chair's Introduction—11:00

Invited Paper

11:05

3aUWb1. Acoustical oceanography with a single hydrophone: Propagation, physics-based processing and applications. Julien Bonnel (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 11, Woods Hole, MA 02543-1050, jbonnel@whoi.edu)

Lobsters, whales and submarines have little in common. Except that they produce low-frequency sound, like many other marine occupants that use sound for communication, foraging, navigation and other purposes. However, unraveling and using the underwater cacophony is not at all simple. This is particularly true for low-frequency ($f < 500$ Hz) propagation in coastal water (water depth $D < 200$ m), because the environment acts as a dispersive waveguide: the acoustic field is described by a set of modes that propagate with frequency-dependent speeds. In this context, to extract relevant information from acoustic recording, one needs to understand the propagation and to use physics-based processing. In this tutorial-like presentation, we will show how to analyze low-frequency data recorded on a single hydrophone. We will notably review modal propagation and time-frequency analysis. We will then show how those can be combined into a non-linear signal processing method dedicated to extract modal information from single receiver, and how such information can be used to localize sound sources and/or characterize the oceanic environment. The whole method will be illustrated on several experimental examples, including geoacoustic inversion on the New England Mud Patch and baleen whale localization in the Arctic.

Exhibit

An instrument and equipment exhibition will be located in the Ballroom near the registration area and meeting rooms.

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

Monday, 2 December, 5:30 p.m. to 7:00 p.m.: Exhibit Opening Reception including lite snacks and a complimentary drink.

Tuesday, 3 December, 9:00 a.m. to 5:00 p.m.: Exhibit Open Hours including a.m. and p.m. breaks serving coffee and soft drinks

Wednesday: 4 December, 9:00 a.m. to 12:00 noon: Exhibit Open Hours including an a.m. break serving coffee.

Session 3pAA

Architectural Acoustics: Assembly Space Renovation Challenges II

Joseph A. Keefe, Cochair
Ostergaard Acoustical Associates, 1460 US Highway 9 North, Ste. 209, Woodbridge, New Jersey 07052

David Manley, Cochair
DLR Group, 6457 Frances St., Omaha, Nebraska 68106

Chair's Introduction—1:00

Invited Papers

1:05

3pAA1. Renovation of a college gymnasium into a multimedia maker-space. Jennifer Nelson Smid (Threshold Acoust., 141 W Jackson Blvd, Ste. 2080, Chicago, IL 60604, jnel04gt@gmail.com) and Gregory A. Miller (Threshold Acoust., Chicago, IL)

The Haverford College Visual Culture Arts and Media (VCAM) project is an adaptive-reuse of an old gymnasium building. The building is stone construction, built in 1900, and unused for decades. The redevelopment included 25 000 SF with a wide range of collaborative, classroom, and laboratory spaces that support the creation, exploration, and study of visual and media arts. The main large volume of the original gymnasium became a lounge and presentation space for multi-media artwork. Significant portions of the original gym floor were removed, allowing a three-story box to be inserted into the volume to house (from top to bottom): a Media, Film Production and Object Study Room; a Film and Media Edit Room; and a 65-seat Screening Room. An original suspended running track was repurposed into flexible seating and circulation to offices and studios. Many acoustic challenges arose from repurposing the large volume gymnasium including reverberation control, structure borne noise and vibration isolation of stacked studios, and acoustic isolation of a noise sensitive screening room located in vicinity of maker spaces utilizing drills, buzz saws, and laser cutters. Audio/Video systems supported a wide range of activities, including a number of sophisticated recording, filming, editing and playback scenarios.

1:25

3pAA2. Curtis Institute of Music—Organ studio renovation. Dan Clayton (Clayton Acoust. Group, 2 Wykagyl Rd., Carmel, NY 10512-6224, danclayton@claytonacoustics.com)

Inspection of former percussion studios in the Curtis mansion basement revealed a warren of four small rooms more like dungeon cells than a future organ studio. Acoustic tile ceilings hid a hodge-podge of noisy air-handling equipment, plus a hundred-year accumulation of pipes, conduit and cables. The client's initial plans for a "paint and carpet" refresh quickly evolved into a major renovation, with a particular design challenge presented by three massive brick piers dividing the studio into four distinct spaces. The largest room became the teaching studio with organ console, a middle room the organ chamber, an internal corridor serves as a sound expansion space for the chamber, and the smallest room holds noisy mechanical equipment and organ blower. Sound isolation ceilings prevent "noise" from traveling up to meeting and reception space above. A lower, finished studio ceiling hides building infrastructure from view, and also functions as a sound distribution space connecting organ chamber to grilles beside and behind the console. Walls have diffuser panels to scatter sound and mid/low-frequency absorber panels to prevent boominess and excessive loudness. The small studio sounds much larger than its actual size, with a pleasant sense of spaciousness and envelopment for organist and listeners.

1:45

3pAA3. Creative reuse and renovation in educational performance environments. Dana Hougland (Shen Milsom & Wlike, LLC, 1822 Blake St., Ste. 2A, Denver, CO 80202, dhougland@smwllc.com)

Educational institutions for the teaching and training students in the arts are often hampered by facitlity limitations. Shifting priotities for the space, outdated acoustical design direction, and budget restrictions place creative constraints on artists and educators. The talent of the performers and arts programs often far exceed the facilities. Funding for renovation is a slow process and may take decades before funds are available to modify and improve the acoustics of performance spaces particularly in rural areas. Volume, shaping, and poor reflected energy are the most common issues. Budget is a limiting factor at many institutions, however, creative collaborations with the design architects allow for significant acoustical improvement. Several case studies will be presented that examine improving the acoustical performance for both the performers and audience while working within the constraints of existing structures. Examples presented include conversion of a classic fan shaped theatre into a musical performance space, conversion of gymnasiums into rehearsal/performance spaces and acoustic enhancement of traditional auditoriums.

2:05

3pAA4. Small town factory into modern multi-purpose music hall. David Manley (DLR Group, 6457 Frances St., Omaha, NE 68106, dmanley@dlrgroup.com) and Rolando De La Cruz (DLR Group, Los Angeles, CA)

The Murphy Arts District in El Dorado, Arkansas is the redevelopment of historic and unique community assets to re-brand the community as a cultural performance mecca and to revitalize the local economy. The first completed phase of the project includes transformation of the historic two-level Griffin Automotive Building into a restaurant and indoor music venue. Acoustic challenges included renovation of a very large volume former factory space into a flat floor music venue to accommodate events from touring rock amplified concerts to wedding banquets. Sound isolation between the music hall and the attached restaurant also presented challenges with historic building constructions.

Contributed Papers

2:25

3pAA5. A method to virtually extend reverberation time of measured impulse responses without losing room coloration. Lévy LeBlanc (Mech. Eng.-Acoust., Université de Sherbrooke, 3-1285 Rue Fabre, Sherbrooke, PQ J1H4W5, Canada, levy.leblanc@usherbrooke.ca), Philippe-Aubert Gauthier, and Alain Berry (Mech. Eng.-Acoust., Université de Sherbrooke, Sherbrooke, PQ, Canada)

For various audio and acoustical applications, it is useful to modify the reverberation time of a measured room impulse response. For example, it can be interesting for creating various modified impulse responses for convolution reverbs in video games or sound designs. With this in mind, it is proposed to modify an impulse response reverberation time while keeping the other reverberation characteristics, such as the frequency response, unaffected. Knowing how a typical impulse response is related to the room characteristics, we propose a new method. First, the impulse response of a room is measured. Then, the reverberation time is evaluated. This reverberation time is next approximated by Sabine-Eyring equation, resulting in an absorption coefficient. A new absorption coefficient is then manually provided to reach a new target reverberation time. Finally, the late reverberation of the original impulse response is convolved with a gaussian white noise to the desired length and stitched with appropriate decaying envelopes to the end of the initially measured impulse response. This paper illustrates that the method works based on simulations and that the modified impulse response is realistic and can be related to the actual room.

2:40

3pAA6. Audiovisual system design and content creation for immersive virtual environments with deployable panoramic display. Mincong Huang (Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, huangm5@rpi.edu), Samuel Chabot, Joseph Wetzel, Johannes Goebel, Hui Su, and Jonas Braasch (Rensselaer Polytechnic Inst., Troy, NY)

State-of-the-art schemata of immersive audiovisual system design mostly rely on *in-situ* stand-up construction with footings and rigid structural supports, an approach limited by low mobility and long set-up time. In this work, a new concept of audiovisual system design for a collaborative Immersive Virtual Environment with flexible and deployable projection elements and modular assemblies, is proposed. Drawing on stand-up configuration from Rensselaer's Collaborative-Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab), a foundationless rectangular panoramic display with round corners is used, incorporating a motorized roll-up framework with mountable fillets. This set-up is then accompanied by a unitized 60-channel Wave Field Synthesis (WFS) linear loudspeaker array. The proposed audiovisual system calibrates the spatial audiovisual rendering by an integrated use of game-engine-based 3-D virtual environments (made in Unity and Unreal) and Max/MSP-based sonification utilities. In particular, an equirectangular transform is applied in virtual cameras and render textures to remove distortion effects from screen geometry. This transform is shared with the WFS array for a congruent presentation of audiovisual content.

Session 3pABa

Animal Bioacoustics and ASA Committee on Standards: Standards in Animal Bioacoustics—Purpose, Need, and Application

Dorian S. Houser, Cochair
National Marine Mammal Foundation, 2240 Shelter Island Drive, San Diego, California 92065

Kurt M. Fristrup, Cochair
Natural Sounds and Night Skies Division, National Park Service, 1201 Oakridge Drive, Suite 100, Fort Collins, Colorado 80525

Invited Papers

1:00

3pABa1. Developing standard exposure-response relationships to inform protected area management. Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, kurt_fristrup@nps.gov)

The "Schultz Curve," which was embodied in ANSI Standard S12.9 Part 4, has been a central informative tool for urban planning and noise impact analyses. Numerous field studies have been conducted by the Volpe Transportation Center and others to develop the foundation for the same type of tool for national parks. I will review what these studies have revealed, and how they compare with the urban noise standard. Park settings vary more widely than urban environments, so one challenge for devising a standard is addressing the potential variety of receptors and the contexts in which effects could be analyzed. The complexity of this issue increases dramatically when considering the nonhuman receptors of noise in national parks, and the effects of noise on human-wildlife interactions.

1:20

3pABa2. Achieving consensus and convergence on a towed array passive acoustic monitoring standard for marine mammal monitoring. Aaron Thode (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9500 Gilman Dr., MC 0238, La Jolla, CA 92093-0238, athode@ucsd.edu) and Shane Guan (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC)

Towed passive acoustic monitoring (PAM) arrays from surface vessels has become a mature technology for marine mammal monitoring during marine seismic surveys. Multiple U.S. federal agencies have expressed a desire for consistent standards when implementing towed PAM. Two workshops involving academics, commercial PAM operators, and regulators have led to a comprehensive outline for an ASA towed array draft standard that covers requirements for initial operation planning, hardware, real-time monitoring and data acquisition, localization procedures, and post-operation validation. It does not cover operational mitigation decision criteria (such as power-down and/or shutdown of seismic airgun arrays), sound source verification, or how to initially establish the required detection range of the system. The standard's fundamental goal is to reduce situations where background noise levels (arising from a variety of mechanisms) prevent effective PAM, and the dominant strategy employed is to standardize how acoustic measurements are logged, reported, and evaluated. Standardized plots would permit a non-technical supervisor or regulator to easily determine whether the PAM operation was self-noise limited under multiple operational circumstances. The approach taken here can serve as a template for future PAM standards for other deployment platforms [Work sponsored by BSSE and ONR Living Marine Resources Program.]

1:40

3pABa3. Working towards a standardized approach to reporting of sound measurements in wildlife studies. Megan F. McKenna (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CO, megan_f_mckenna@nps.gov) and Kurt M. Fristrup (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

We present guidance we have developed and adapted through attempts to synthesize and distill the diverse literature documenting wildlife responses to noise into concise support for conservation planning. Our systematic review revealed significant inconsistencies in the use of—and specifications for—acoustic metrics. Studies typically report noise levels in decibels (dB), yet these values can arise from a diverse array of methods and processing procedures. Though some of this diversity reflects varied objectives and acoustical contexts, we believe that normative guidance and better awareness of standards would enhance the interpretation, repeatability of published studies, and incorporation of results into conservation planning. Nested within this larger issue is the vexing problem of calibration, especially in relation to consumer audio recorders or emerging products for bioacoustical monitoring. While it is possible to extract useful information about acoustic phenology or species presence from uncalibrated audio recordings, calibration can deliver a much broader range of information. We review the suite of calibration options presently available, and our experience with implementing and sharing solutions.

2:00

3pABa4. Standard for acoustic metadata for passive acoustic monitoring. Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sbaumann@ucsd.edu) and Marie A. Roch (Dept. of Comput. Sci., San Diego State Univ., San Diego, CA)

Passive acoustic monitoring often relies on data collected with various instrumentation, by multiple research groups in different locations, under diverse conditions, and with a variety of data analysis methods. To make the output comparable and to preserve data for long time-series analysis, metadata needs to be reported in a standardized way. This will enable broader metadata studies and the sharing of data in repositories, which will allow for cross fertilization of analyses and hence cost savings for new projects. The Acoustical Society of America Working Group S3-SC1-WG7 "Acoustic Metadata for Passive Acoustic Monitoring" is in the process of defining this standard. We are building upon schemata developed for the metadata database software "Tethys" [Roch *et al*., *Ecol. Inf.* **31**, 122–126 (2016)]. Instrument deployments and on- or off-effort detections of acoustic signals were defined within this manuscript. The current deployment schema is being implemented in the National Center for Environmental Information Passive Acoustic Monitoring Archive. We have expanded the vocabulary to include mobile platforms, vector sensors, localizations, and calibrations. Exemplary scenarios were developed to test the comprehensiveness of the current standard definitions for these frameworks. We also consider extensibility, making sure that the standard is flexible enough to support new ideas.

2:20–2:35 Break

2:35

3pABa5. Establishment of ANSI/ASA S3/SC1.6—Standardization of evoked potential hearing test methods in toothed whales. Dorian S. Houser (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org)

The use of auditory evoked potentials (AEPs) to test the hearing of toothed whales has produced hundreds of toothed whale audiograms over the last two decades. Unfortunately, multiple approaches to obtaining AEP audiograms have been employed, including different methods for estimating the hearing threshold, the use of different acoustic test stimuli, different approaches to calibration, etc. Lack of methodological consistency has resulted in considerable variability in threshold estimates between researchers and laboratories and conclusions about differences in the hearing abilities of toothed whale populations that are likely erroneous. To address this issue, ANSI/ASA S3/SC1.6-2018 was developed. The document proposes standardized methods for performing AEP hearing tests in toothed whales. Implementation of the standard should reduce variability in threshold estimates obtained by different laboratories and researchers and make the resulting audiograms more comparable. Although AEP hearing tests are common for the clinical assessment of hearing in stranded whales and their use for this purpose is not substantially impacted by methodological variances, audiogram comparability is necessary for the broader implementation of AEP audiograms into environmental impact assessments and regulation. Thus, the standard should facilitate environmental stewardship efforts to manage the impact of anthropogenic sound on marine mammals.

Contributed Papers

2:55

3pABa6. Explosions recorded underwater offshore of southern California. Sean M. Wiggins (Scripps Inst. of Oceanogr., 9500 Gilman Dr. La Jolla, La Jolla, CA 92093-0205, swiggins@ucsd.edu), Anna Krumpel (Eberhard Karls Univ. of Tuebingen, Tuebingen, Germany), Macey Rafter, LeRoy Dorman, John Hildebrand, and Simone Baumann-Pickering (Scripps Inst. of Oceanogr., La Jolla, CA)

Sounds from explosions can travel long distances underwater due to their high sound pressure levels, potentially impacting marine life. We evaluated and characterized explosion sounds from underwater recordings to better understand the signature of these sounds, how they propagate in the ocean, and how they may be differentiated for different source types. A controlled experiment using small ($\sim$2 g) "seal bomb" sources at various source-receiver ranges was conducted offshore of southern California to characterize their waveform signature, including bubble pulses due to subsurface explosions $\sim$4 m depth and peak source level estimate of 234 dB re 1 μPa at 1 m. Opportunistic recordings of missile explosions during military exercises offshore of southern California were made by autonomous hydrophone recorders $\sim$1000 m deep. The military explosions were presumably at or near the sea surface and did not exhibit bubble pulses, allowing them to be differentiated from fisheries seal bomb sources for manual or automatic signal classification in underwater acoustic recordings.

3:10

3pABa7. Damage potential index: A single numeric expression for noise exposure impacts. Michael A. Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@OCR.org)

Regulatory noise exposure thresholds are currently based exclusively on exposure amplitude. These thresholds were only recently bracketed in the frequency domain by auditory bands of marine mammals in a set of "M weighting" curves. While these curves are an improvement on understanding and applying regulatory thresholds for "Level B" behavioral disturbance exposures, the suite of regulatory guidelines still falls short of expressing actual damage or disruption potential of any particular noise exposure. A correlation between signal kurtosis (of amplitude variability over time) and hearing damage has been well established – with overall signal amplitude remaining an important variable. Understanding that regulators prefer single numeric "go/no go" thresholds has likely been a factor in not adopting any more nuanced expressions of exposure damage potential. We are proposing a single numeric that integrates both amplitude and kurtosis in the time/frequency domain as a "damage potential index" which would more accurately express the impacts of a disturbing or damaging sound.

3:25

3pABa8. Accessing standardized acoustic metadata: A web-based graphical interface for an acoustic metadata server. Jeffery Cavner (SDSU, 5500 Campanile Dr., San Diego, CA 92182, jeff.cavner@gmail.com), Marie Roch, Roger Whitney (SDSU, San Diego, CA), and Simone Baumann-Pickering (Scripps Inst. of Oceanogr., La Jolla, CA)

The Tethys acoustic metadata workbench consists of a set of vocabulary terms designed to provide a standard method to represent detections,

3p WED. PM

classifications, and localizations of biological, ambient, and anthropogenic signals in acoustic data; and a database that permits users to archive and manipulate these data as well as access contextual data such as ephemeris, sea surface temperature, etc. We present a new graphical web client for Tethys that adds a robust set of tools for extensive data exploration in a visualization environment that includes maps and data conversion utilities along with the ability to save queries to provide repeatable results. The web interface provides the ability to query and visualize acoustic data from a web browser with little prior experience. A graphical interface permits the generation of queries without knowledge of the underlying XQuery database language. The standard mode is designed for users with limited knowledge of the Tethys schemata to obtain basic information, see results on a map, and save data in a variety of formats such as Matlab, CSV, and R. Advanced users can switch to a more comprehensive interface that lets them construct and share more sophisticated queries for general use.

WEDNESDAY AFTERNOON, 4 DECEMBER 2019 CROWN, 1:00 P.M. TO 1:45 P.M.

Session 3pABb

Animal Bioacoustics: Animal Bioacoustics Poster Session

Matt Schalles, Chair
Biomedical Engineering, Boston University, Department of Cognitive & Neural Systems, 677 Beacon Street, Boston, Massachusetts 02215

All posters will be on display and all authors will be at their posters from 1:00 p.m. to 1:45 p.m.

Contributed Papers

3pABb1. Laboratory rats (*Rattus norwegicus*) detect the Fanssen effect. Alayna L. Collins (Psych., Univ. of Nebraska-Kearney, Kearney, NE) and Evan M. Hill (Psych., Univ. of Nebraska-Kearney, 2507 11th Ave., Kearney, NE 68849, hillem@unk.edu)

The purpose of this study was to determine whether laboratory rats (*Rattus norwegicus*) were able to detect the Franssen Effect (FE). The FE is elicited by simultaneously presenting a spatially split signal to the left and right of the listener. One side presents a transient tone with rapid linear offset and the opposite presents a sustained tone with a ramped onset. The listener perceives a single, sustained tone localized on the side of the transient signal. Previous research demonstrated perception of the FE in three mammalian species by monitoring gaze-shift via the scleral search coil technique. However, this method requires placing contact lenses in each eye and restraining non-human subjects, making it suboptimal for use in many species. Alternatively, two species of bird were shown to detect the FE using an operant behavioral method. The current study used a similar two-choice procedure to determine detection of the FE in octave steps, ranging from 1–32 kHz. The results of this study found that laboratory rats also detect the FE. Human participants also demonstrated detection of the FE using this behavioral technique at rates comparable to the previous literature. Therefore, this behavioral methodology is a viable option for testing this auditory phenomenon.

3pABb2. Design of a miniature evoked potential testing system for animal assessment of hearing. Al Yonovitz (The Univ. of Montana, Dept. of Communicative Sci. and Disord., Missoula, MT 59812, al.yonovitz@umontana.edu) and Silas Smith (The Univ. of Montana, Missoula, MT)

A specially built apparatus has been designed to allow novel methodologies for simultaneous measurement of early, middle and late evoked potentials. Evoked potentials can be measured in active, unsedated, unrestrained rats. Typically, early evoked potentials are measured with animals that are anesthetized. Middle and late evoked potentials must be measured in awake animals. Chronically implanted electrodes provide the bioelectric potentials. To accomplish this active measurement, data are transferred wirelessly from a portable unit attached to the rat, to a computer via wireless Bluetooth. The system is controlled with a Raspberry Pi. Included are a differential input amplifier, A/D converter and low and high pass filters. The miniature system also includes a means to provide a wide range of auditory stimuli typically driving a small insert earphone. All programs and auditory stimuli are stored on micro SD cards. The pack stays in place and is not removable by the animal. This system is ideally suited for measurements of backward masking middle latency responses as a marker for Auditory Processing Disorders (APD). The pack is light and easily attachable.

3pABb3. Long-term exposure to substrate-borne vibrations and the behavioral response of field crickets (Gryllidae). Jessica Briggs (Dept. of Biological Sci., Univ. of New Hampshire, 6 Stark Ave., Dover, NH 03820, jb1424@wildcats.unh.edu) and Daniel R. Howard (Dept. of Biological Sci., Univ. of New Hampshire, Durham, NH)

Anthropogenic noise is a present in most landscapes, with a variety of impacts documented across taxa. Much of the research investigating the impact of anthropogenic noise on animals in terrestrial ecosystems has focused primarily on airborne sound, with little attention to substrate borne vibrations. Many invertebrates rely on substrate borne signals and cues to communicate and detect predators or prey, but little is understood about the impact of substrate-borne vibrational noise on invertebrate behavior. We designed an experiment to determine if wild populations of field crickets (*Gryllus pennsylvanicus and G. veletis*) that occupy habitats close to heavily trafficked roadways typified by high levels of substrate-borne vibrational noise displayed a lower response to substrate-borne cues compared to populations in quiet locations. Low frequency substrate-borne vibrational stimuli (100–1000 Hz bandpass brown noise) was played back to singing males in the field to determine behavioral thresholds to vibration. Populations at roadside sites were exposed to substrate vibrations of 0.36–0.40 mm/s peak to peak while crickets at quiet sites were exposed to 0.29–0.30 mm/s peak to peak. Behavioral threshold responses of each population are presented.

3pABb4. Evaluation of cochlear compression in a bottlenose dolphin: A forward-masking evoked-potential study. Vladimir Popov (Inst. of Ecology and Evolution, 33 Leninskij prosp., Moscow, 119071, Russian Federation, popov.vl.vl@gmail.com), Alexander Supin, Dmitry Nechaev, and Evgeniya Sysueva (Inst. of Ecology and Evolution, Moscow, Russian Federation)

Compressive nonlinearity allows to analyze sounds within a very wide level range. It is a property of the "active" cochlear mechanism based on electromotility of outer hair cells. It was not known is this mechanism capable to function at high frequencies accessible for hearing of odontocetes. Nonlinearity can be measured assuming that on-frequency masking is subjected to compression whereas the low-frequency masking is not. To eliminate confounding effects (lateral suppression and off-frequency listening), forward masking of near-threshold test stimuli must be used. In a bottlenose dolphin, on- and low-frequency forward masking of near-threshold test stimuli was measured in conjunction with the auditory evoked potential method. Test stimuli were short pips of 90 kHz frequency, the maskers were pips of 90 kHz (on-frequency), 76, 64, or 54 kHz (0.25, 0.5, and 0.75 oct below the test). Masker-to-signal delays varied from 2 to 20 ms. With increasing the delay from 5 to 20 ms, the on-frequency masker level at threshold (MLT) increased by 27 dB, whereas the low-frequency MLT increased by 4.5 dB. The compression was evaluated as 1:6. [Work supported by Russian Foundation for Basic research, Grant 18-04-00088 to VP.]

3pABb5. Additional data on the lower jaw morphology in odontocetes: Panbone thickness. Evgeniya Sysueva (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, evgeniasysueva@gmail.com) and Dmitry Nechaev (Inst. of Ecology and Evolution, Moscow, Russian Federation)

The mandibular hypothesis by Norris (1968, 1969) mandible implies that in odontocetes, sounds pass to the ear through a mandibular window (panbone) where the bone is fairly thin and the intra-mandibular fat body. However, thickness of the bone depends on size of the animal. In small and medium-sized odontocetes it is 0.57 to 2 mm, whereas in large ones, such as the killer whale, it attains 5.48 mm (Nummela *et al*., 2007), which may worsen the conditions for sound penetration into the fatty body. Despite the thick panbone region, the hearing of the killer whale is not inferior to that of smaller odontocetes. These facts encouraged us to obtain additional data on the lower jaw morphology. We 3-D-scanned and analyzed 19 mandibles from adult odontocetes of various species: the porpoises, Irrawaddy dolphin, common dolphin, bottlenose dolphin, beluga, and killer whale. The panbone thickness was measured in all the samples. Results of the study did not reveal a direct correlation of the body size and panbone thickness. Further, the obtained data will be useful for making a mathematical model of sound conduction dependence on the panbone thickness. [Work supported by the Russian Science Foundation, Grant # 17-74-20107 to E.V.S.]

Session 3pAO

Acoustical Oceanography and Animal Bioacoustics: Bioacoustics and Acoustical Oceanography: 20 Years Later II

Andone C. Lavery, Cochair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 98 Water Street, MS 11, Bigelow 211, Woods Hole, Massachusetts 02543

Kelly J. Benoit-Bird, Cochair
Monterey Bay Aquarium Research Institute, 7700 Sandholdt Road, Moss Landing, California 95039

Contributed Papers

1:00

3pAO1. Monitoring transitions between dispersed and school modes of sardines and anchovies with Bioacoustic Absorption Spectroscopy (BAS) measurements. Orest Diachok (Poseidon Sound, 11100 Johns Hopkins Rd., Laurel, MD 20723, orestdia@aol.com)

BAS measurements at a biologically intense site in the Santa Barbara Channel permitted monitoring the temporal variation of absorption lines associated with fish in dispersed and school modes. Transmission Loss (TL) was measured at frequencies between 0.25 and 5 kHz between a fixed source and fixed receiver array separated by 3.7 km for 20 hours. Echo-sounder measurements: the average depth of fish at night was about 13 m. Trawls: 16 cm sardines and 10.5 cm anchovies dominated. The frequency dependence of bio-attenuation was derived through comparison of TL measurements and calculations. The latter were based on a normal mode code recently extended for broadband calculations (Porter, 2019). Bio-attenuation at 1.1 kHz, the resonance frequency of dispersed sardines, was 35 dB at 2200 l, 15 dB at 0100 l and 35 dB at 0400 l. When bio-attenuation at 1.1 kHz was minimum, bio-attenuation 0.4 kHz, attributed to schools of sardines, was maximum. The absorption line at 0.4 kHz is consistent with previously reported measurements and theoretical calculations of the resonance frequency of schools of 16 cm sardines (Diachok, 1999). Bio-attenuation at 1.8 and 0.6 kHz, attributed to 10.5 cm anchovies in dispersed and school modes respectively, were also anti-correlated. [Research supported by ONR.]

1:15

3pAO2. Variability of the geoacoustic properties of infauna-bearing marine sediments. Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu), Megan S. Ballard (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Kelly M. Dorgan (Dauphin Island Sea Lab, Dauphin Island, AL), Gabriel R. Venegas, Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Infauna influence sediment physical parameters, such as porosity, grain size, and pore fluid properties, through bioturbation and other biological processes. To investigate the effects of the resulting sediment inhomogeneity, *in situ* measurements of compressional and shear wave propagation were acquired in Petit Bois Pass, west of Dauphin Island, Alabama, USA. After completion of the acoustic measurements, diver core samples were collected from between the sources and receivers to characterize the physical properties of the seabed along the acoustic propagation paths. The collected sediments were analyzed for infauna biomass and geotechnical properties to support acoustic propagation modeling. The acoustic data are analyzed in the context of infaunal functional groups, which are based on acoustically significant traits such as body type (hard/soft and large/small) and infaunal activity (sediment mixing or tube building). While existing sediment acoustics models do not explicitly account for effects of biology, they do allow for parameterization of various physical properties, 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 gain insight into the acoustical effects of infauna. [Work supported by ONR.]

1:30

3pAO3. Combined passive acoustic and video monitoring of coral reefs to better understand reef soundscapes and multi-species interactions. Lauren Freeman (NUWC Newport, Newport, RI, lauren.a.freeman@navy.mil), Simon Freeman (NUWC Newport, Newport, RI), Paul Gader, Ronald Fick, Nicholas Kroeger (Univ. of Florida, Gainesville, FL), Aaron Thode (SIO, UCSD, La Jolla, CA), Kayla Thilges (Ocean Eng., Univ. of Rhode Island, Saunderstown, RI), Lora J. Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Sonia Rowley (ARL, Univ. of Hawaii, Honolulu, HI), Jeffrey Schindall (US Naval Res. Lab, Washington, DC), and Alexis Johnson (NUWC Newport, Newport, RI)

A detailed understanding of ecological soundscapes provides a window into ecosystem state that might otherwise go undetected. Rather than consider acoustic emissions of a particular species, a top down ecosystem acoustics point of view gives potentially valuable insights to reef ecosystem function. Simultaneous passive acoustic and video data were collected from coral reefs off the Kona Coast of Hawaii in January 2019 during a soundscape experiment. Frequent interactions between fish species as well as collective behaviors are identified and tracked by applying computer vision to video data. Comparison relating video data to acoustic data allows assessment of the same behaviors as simultaneously reflected in directional passive acoustic data. Collective consideration of acoustics and video offers insight into acoustic signatures of particular soniferous communities and behaviors amongst and between reef species. Machine learning applications match patterns between fish trajectories in videos and the sound field as received by a directional antenna. The relative level of biological activity and presence or absence of particular behaviors provides key data in the effort to understand coral reef ecosystem state.

1:45–2:00 Break

2:00

3pAO4. Sounds from airguns and blue whales recorded from a long term hydrophone network in the Southern Indian Ocean. Maëlle Torterotot (Laboratoire GéoSci. Océan, IUEM, Univ. of Brest & CNRS, Plouzané, France), Flore Samaran (Lab-STICC UMR CNRS 6285, Brest, France, flore.samaran@ensta-bretagne.fr), and Jean-Yves Royer (Laboratoire GéoSci. Océan, IUEM, Univ. of Brest & CNRS, Plouzané, France)

The Southern Indian Ocean is a seasonal habitat for different blue whale sub species and sub populations. Their presence has been demonstrated by continuous acoustic records from the OHASISBIO hydrophone network, deployed since 2010 and which comprises 5 to 9 fixed mooring sites spread over a region from 24 deg to 56 deg South and from 52 deg to 83 deg East. Automated detection based on dictionaries and sparse representation has been used to detect and count the different stereotyped or non-stereotyped calls emitted by blue whales and recorded in this long-term dataset. However false alarm rate increased sometimes by the detection of a recurrent interference in the same frequency band of blue whales: the airgun shots produce during surveys for oil and gas located somewhere in the Indian Ocean and beyond. This finding allows us to characterize the seasonal and interannual variability in airgun sounds in this blue whale habitat. This study reinforce the fact that anthropogenic noise could be detected at large scale in pelagic environment and may affect marine mammal specially species who used low frequency bands.

2:15

3pAO5. Potential impact of mid-frequency active sonar on whales from passive acoustic monitoring data. Alba Solsona Berga (Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0205, asolsonaberga@ucsd.edu), Jennifer S. Trickey, Ally Rice (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Ana Širović (Texas A&M Univ. Galveston, Galveston, TX), Marie A. Roch (San Diego State Univ., San Diego, CA), Charles G. Paxton, Cornelia S. Oedekoven (Univ. of St. Andrews, St. Andrews, United Kingdom), Sean M. Wiggins, John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Len Thomas (Univ. of St. Andrews, St Andrews, United Kingdom), and Simone Baumann-Pickering (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Behavioral response studies of tagged whales have documented an adverse reaction to mid-frequency active sonar (MFAS). The relationship between MFAS and the acoustic behavior of whales is complex and requires accounting for natural variability in species presence. We examine the acoustic response of Cuvier's beaked whales (*Ziphius cavirostris*) and blue whales (*Balaenoptera musculus*) from long-term acoustic recordings at three sites in Southern California waters from 2009 to 2015. Presence and absence were marked in 1-min bins for all signal types. Generalized estimating equations (GEEs) were used to model relationships between species acoustic detection and temporal and sonar covariates while accounting for residual temporal autocorrelation. Both species showed considerable temporal variability in calling. Modeling at one site showed decreased detections of beaked whale echolocation clicks when sonar signals had higher received levels at the recorders and when the proportion of bins containing sonar increased. On cessation of sonar signals, beaked whale detection increased over periods of up to about a week. Analyses of blue whale response and a comparison of multiple sites will also be presented. This work represents substantial progress in our understanding of the impact of MFAS on cetaceans, which is crucial for effective context-dependent management of anthropogenic activities.

2:30

3pAO6. Cuvier's beaked whale tracks in Southern California. Eric Snyder (Scripps Inst. of Oceanogr., Univ. of California, San Diego, 9176 Regents Rd. Apt. A, La Jolla, CA 92037, e5snyder@ucsd.edu), Sean M. Wiggins (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Simone Baumann-Pickering, and John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Cuvier's beaked whales (*Ziphius cavirostris*) are difficult to study due to infrequent sightings and their deep-diving behavior. One approach to studying their behavior is to use arrays of hydrophones to localize and track individual beaked whales. We collected over one year of small aperture (~1 m) broadband acoustic array recordings at different sites offshore of the coast of Southern California, a region with a considerable presence of Cuvier's beaked whales. We use the time difference of arrival between hydrophone pairs in our arrays to determine the direction of an echolocating beaked whale. When the source directions are cross-fixed from multiple arrays, the location of an animal can be determined and sequential locations can be used to form tracks. Tracks were used to determine a number of signal characteristics, such as source sound pressure level, beam directionality, and click temporal patterns as well as a number of animal behaviors including swim speed, group size, descent angle, and foraging depth.

2:45

3pAO7. Passive acoustic monitoring to assess depredation of killer whales on demersal longlines during soaking. Gaëtan Richard (ENSTA Bretagne, 2 rue François Verny, Brest 29200, France, gaetan.richard@ensta-bretagne.fr), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), Christophe Guinet (Ctr. d'Etudes Biologiques de Chizé, Villiers-en-Bois, France), and Flore Samaran (ENSTA Bretagne, Brest, France)

Toothed whales feeding on fish caught on longlines is a growing issue worldwide. This issue named depredation has a serious socio-economic impact and raise conservation questions. In the French Patagonian toothfish fishery (Southern Ocean), a good understanding of the depredation behavior by killer whales is urgently needed to find adequate mitigation solutions. However, depredation on demersal longlines has been historically assessed from surface observation during hauling phases, thus the underwater dimension of depredation is not understood. Here, we used passive acoustic monitoring to record echolocation activity of killer whales and fishing vessels' acoustic signatures during the fishing activity. Fixed hydrophone deployed on soaking longlines without any fishing activity nearby has recorded echolocation sounds. These recordings suggest that killer whales were feeding within the vicinity of longlines and thus very likely depredating on soaking longlines. Additionally, we also investigated fishing vessels' acoustic signatures. We observed differences of acoustic cues emitted by the vessels between the setting and hauling phases. This suggests that killer whales may be able to recognize the boat activity and therefore localize longline positions to depredate.

3:00

3pAO8. Identification of species-specific delphinid echolocation click types. Rebecca Cohen (Scripps Inst. of Oceanogr., Univ. of California, San Diego, Ritter Hall 101, 8635 Kennel Way, La Jolla, CA 92037, rec022@ucsd.edu), Amanda Leu, Kaitlin E. Frasier (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA), Melissa Soldevilla (NOAA Fisheries, Southeast Fisheries Sci. Ctr., Miami, FL), and John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

Delphinid echolocation clicks are often recorded in large, high-quality datasets collected by moored autonomous passive acoustic sensors, but the particular species present during these acoustic encounters is generally unknown. We combined two acoustic data types with visual observations to obtain species identity for some of these clicks. We assigned species identity to acoustic encounters in eight years of recordings at five western North Atlantic mooring sites using sighting data from contemporaneous shipboard and aerial visual surveys. We also assigned species identity to acoustic encounters in towed acoustic array data collected concurrent with visual surveys during four ship-based cetacean surveys in the western North Atlantic and the Gulf of Mexico. 114 mooring encounters with seven species and 181 towed array encounters with nine species were labeled this way. We identified recurring click types during all encounters using an unsupervised clustering algorithm to groups clicks based on pairwise spectral distances. Labeled click types thus identified in the moored data were compared to labeled click types identified in the towed array data to determine their consistency and utility for delphinid species discrimination in moored passive acoustic data. Results for bottlenose dolphin, Atlantic spotted dolphin, Risso's dolphin, and pilot whales will be presented.

Session 3pBA

Biomedical Acoustics: Biomedical Acoustics Poster Session

Alessandro Ramalli, Chair
Cardiovascular Imaging and Dynamics, KU Leuven, UZ Herestraat 49 - Box 7003, Leuven 3000, Belgium

All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Contributed Papers

3pBA1. Measurements of the speed of sound and attenuation of healthy and malignant prostate cells. Maria-Teresa Herd (Biological and Physical Sci., Assumption College, 50 College Ave., South Hadley, MA 01075, therd@mtholyoke.edu), Penelope Taylor, and Amy Longstreth (Mount Holyoke College, South Hadley, MA)

The National Institute of Health reports that for men prostate cancer is the most common cancer and the second most common cause of death due to cancer. The most recent data show that there are about 186 new cases of prostate cancer per 100,000 men, with 24 deaths per 100,000 men each year. Prostate cancer is difficult to detect and diagnose. Currently prostate-specific antigen (PSA) levels are tested, and biopsies are performed if the PSA levels are high. This is not a very accurate indicator, since two thirds of all biopsies are benign. This study explores the possibility of using quantitative ultrasound as a diagnostic for prostate cancer, by studying the differences in tissue properties between benign and malignant prostate cells. Speed of sound and attenuation as a function of frequency between 2 and 18 MHz were measured and compared for a cancerous prostate cell line and a healthy prostate cell line. Speed of sound for the cancerous cells line was found to be 1521.4 ± 0.8 m/s, which was equivalent to the speed of sound for the healthy cell line of 1521.5 ± 0.6 m/s. The average attenuation coefficient was 0.091 ± 0.003 dB/(cm MHz) for the cancerous prostate cell line and 0.057 ± 0.003 dB/(cm MHz) for the healthy prostate cell line, showing a higher attenuation for the cancerous cell line.

1:00

3pBA2. Effect of low intensity ultrasound on cancerous prostate cell viability compared to viability of healthy prostate cells. Maria-Teresa Herd (Biological and Physical Sci., Assumption College, 50 College Ave., South Hadley, MA 01075, therd@mtholyoke.edu), Barbara Marquez, Hannah Seay, and Chloe Verducci (Phys., Mount Holyoke College, South Hadley, MA)

Cancer continues to be a leading cause of death in the United States. Currently, the most commonly utilized treatment options in radiotherapy and chemotherapy raise serious biocompatibility and toxicity concerns. Given the adverse side effects of current treatment, it is important to explore modalities that can clearly discriminate between healthy and cancerous tissue. This investigation identifies low intensity ultrasound as a candidate for cancer therapy and tests whether it demonstrates a discriminatory response between cancerous prostate cells and healthy prostate cells. Cells were exposed to a 5 MHz, 0.126 W/cm^2 wave for 60 minutes via a single element unfocused transducer in a water tank-controlled environment at 33 °C. Cell counts (alive and dead) before and after exposure to ultrasound were performed. On average cancerous prostate cells showed a viability of 53% compared to a viability of 99% for benign prostate cells, indicating a significant difference in observed death between healthy and cancerous prostate cells.

3pBA3. Transfer learning for ultrasound tongue contour extraction with different domains. M. Hamed Mozaffari (Elec. and Comput. Eng., Univ. of Ottawa, SITE4009 - 800 King Edward Ave., Ottawa, ON K1N 6N5, Canada, mmoza102@uottawa.ca), David Sankoff (Mathematics and Statistics Dept., Univ. of Ottawa, Ottawa, ON, Canada), and Won-Sook Lee (Elec. and Comput. Eng., Univ. of Ottawa, Ottawa, ON, Canada)

Due to the low-contrast characteristic and noisy nature of ultrasound images, it might require expertise for non-expert users to recognize tongue gestures. In the last few years, deep learning methods have been used for delineating and tracking tongue dorsum. Deep convolutional neural networks (DCNNs), which have shown to be successful in medical image analysis tasks, are typically weak for the same task on different domains. In many cases, DCNNs trained on data acquired with one ultrasound device, do not perform well on data of varying ultrasound device or acquisition protocol. Domain adaptation is an alternative solution for this difficulty by transferring the weights from the model trained on a large annotated legacy dataset to a new model for adapting on another different dataset using fine-tuning. In this study, we addressed the problem of domain adaptation on ultrasound datasets for tongue contour extraction. We trained a U-net network, and then with several surrogate scenarios, some parts of the trained network were fine-tuned on another dataset as the domain-adapted networks. We repeat scenarios from target to source domains to find a balance point for knowledge transfer from source to target and vice versa. The performance of new fine-tuned networks was evaluated on the same task with images from different domains. Our experimental revealed that a deep model can be used effectively for different domains using a knowledge balance point between those models.

3pBA4. BowNet: Dilated convolutional neural network for ultrasound tongue contour extraction. M. Hamed Mozaffari (Elec. and Comput. Eng., Univ. of Ottawa, SITE4009 - 800 King Edward Ave., Ottawa, ON K1N 6N5, Canada, mmoza102@uottawa.ca), David Sankoff (Mathematics and Statistics Dept., Univ. of Ottawa, Ottawa, ON, Canada), and Won-Sook Lee (Elec. and Comput. Eng., Univ. of Ottawa, Ottawa, ON, Canada)

One usage of medical ultrasound imaging is to visualize and characterize human tongue shape and motion during a real-time speech to study healthy or impaired speech production. Due to the low-contrast characteristic and noisy nature of ultrasound images, it might require expertise for non-expert users to recognize tongue gestures in applications such as visual training of a second language. Several end-to-end deep learning segmentation methods provide promising alternatives with higher accuracy and robustness results and without any intervention. Employing the power of the graphics processing unit with state-of-the-art deep neural network models makes it feasible to have new fully automatic, accurate, and robust segmentation methods with the capability of real-time performance. This paper presents a new novel deep neural network for tongue contour extraction, BowNet, benefits from exploitation capability of dilated convolution by

effectively expanding the receptive field without losing resolution to extract clear tongue contours. Also, efficient abstract context exploration is carried out by down-sampling layers to achieve segmentation results with high resolution and relevancy. Two versions, BowNet and wBowNet, are studied qualitatively and quantitatively over datasets from two different ultrasound machines. Our experiment disclosed the outstanding performances of the proposed models in terms of accuracy and robustness in comparison with similar sized models.

3pBA5. Cylindrical ultrasound transducers for intravascular and endolumenal imaging applications. Carl Herickhoff (Radiology, Stanford Univ., 3155 Porter Dr., Palo Alto, CA 94304, carl.herickhoff@stanford.edu), Arsenii Telichko, and Jeremy J. Dahl (Radiology, Stanford Univ., Palo Alto, CA)

Conventional ultrasound imaging transducers are fabricated using planar geometries—of materials and in most subsequent processing (bonding, dicing, etc.); this can lead to complications in assembling catheter- or endoscope-based devices, and/or significant inherent limitations in their design and application. Miniature cylindrical transducer devices offer the potential for more straightforward design for 360-deg imaging applications, albeit with unique fabrication challenges. We have developed a novel technique for processing cylindrical ultrasound transducer geometries, and used it to fabricate prototype devices designed for intravascular and endolumenal applications. We created a custom dicing setup allowing a tube of radially poled piezoceramic (PZT-4) as small as 4.6 Fr (1.53 mm diameter) to be cut circumferentially, to a specified depth or completely through the wall of the tube, at precise axial locations. The resulting cylindrical transducer was connected to a function generator and power amplifier, and driven at 7.5 MHz to generate axisymmetric, radially directed acoustic radiation force impulse (ARFI) displacements up to 19 μm in a tissue-mimicking elasticity phantom. This demonstrates that miniature cylindrical transducers can be made to enable 360-deg elasticity imaging of the mechanical properties and structure of tissues, to identify vulnerable atherosclerotic plaques or quantify softening of the cervix prior to labor onset.

3pBA6. Effect of ultrasound on fibrin clot structure. Sheeny L. Levengood (BTG Int., 19910 North Creek Parkway, Bothell, WA 98021, sheeny.levengood@btgplc.com), Sophia Song, Sarah Beebe, Alex Hannah, Curtis Genstler, and Misty L. Noble-Vranish (BTG Int., Bothell, WA)

A commercially available catheter (EKOS EkoSonic catheter) has been used in the clinic with lytic drugs to clear blood clots associated with pulmonary embolism, deep vein thrombosis and peripheral arterial occlusion. Operating at 2.3 MHz, transducers within the catheter deliver short, variable amplitude pulses at intensities between 3.4 and 11.5 W/cm^2. Lysis enhancement in the presence of ultrasound is non-thermal and generally attributed to increased transport and, therefore, increased efficiency of the lytic. The enhanced lytic effect may also be associated with increased availability of lytic binding sites associated with changes in the clot's fibrin structure. To better understand the specific effects of ultrasound on fibrin networks within blood clots, we utilized scanning electron microscopy (SEM) and clot turbidity measurements to observe and quantify changes in fibrin mesh. SEM images of Factor XIII-depleted fibrinogen clots show a significant decrease ($p < 0.05$) in fibrin fiber diameter in the presence of ultrasound (33.8 $\pm$ 17.2 nm) compared to control (63.4 $\pm$ 24.0 nm). Clot turbidity is reduced following treatment with ultrasound, which correlates with decreased fiber diameter. These results indicate that ultrasound alters fibrin structure, which may contribute to enhanced binding of lytic drug and clot lysis.

3pBA7. On an improvement of the old voice by the reflection of the amplified voice. Seonggeon Bae (School of Software Application, Kangnam Univ., 40, Kangnam-ro, Giheong-gu, Youngin-si, Gyeonggi-do, Korea, Youngin 446-702, South Korea, sgbae@kangnam.ac.kr) and Myungjin Bae (Commun. Eng., Soongsil Univ., Seoul, South Korea)

When people get older, they lose their energy and get tired of everything. At this time, the voice utterance ages and gradually changes to bass, and the sounding frequency of the first resonance frequency decreases with age. Voice phonation is produced by the release of pressure from the lungs through the vocal folds, causing resonance in the vocal tract. In this study, we proposed a new method to improve the aging of vocal organs with their voices using the voice utterance characteristics. We have to hear the vocal organs of the experimenter as usual loudness over a cup that contains the sound recorded voice, the voice-aging properties were measured in one-minute increments. In this study, the average sound pressure level when the aging voice removal was restored to + 6.2dB spectrum of the high-frequency range is improved to the average size + 13.7dB. And the experimenter showed regained the confidence of the utterance to think that his voice clarity improvement over the ages.

3pBA8. Sonogenetics: Non-invasive cellular manipulation using ultrasound. Aditya Vasan (Mech. Eng., UCSD, Structural and Mater. Eng., 320, La Jolla, CA 92092, advasan@eng.ucsd.edu), Uri Magaram, Corinne Lee-Kubli, Yusuf Tufail, Jose Mendoza, Marc Ramirez, Connor Weiss (Salk Inst., San Diego, CA), Sreekanth Chalasani (Salk Inst., La Jolla, CA), and James Friend (Mech. Eng., UCSD, La Jolla, CA)

Optogenetics has been an important tool used in manipulating different cell types, understanding various diseases and has recently been studied as an alternative to electrical manipulation techniques used in a clinical setting. It has the drawback of being an invasive cellular manipulation technique, which limits its applications. Ultrasound is an alternative, non-invasive approach to precisely control cellular function. We propose the development of ultrasound transducers to selectively manipulate cells that have been transfected with mechanically sensitive proteins, in a new technique termed "Sonogenetics." This study describes recent results in single crystal piezoelectric transducer design and characterization, results from *in-vitro* studies to identify mechanosensitive proteins, and an *in-vivo* study where miniature head mounted transducers were used to stimulate the ventral tegmental area of the mouse in order to trigger dopamine release. Ultrasound based stimulation offers superior spatial and temporal control compared to other cellular manipulation techniques. This study enabled development of the tools required for achieving sonogenetic manipulation of cells by presenting novel transducer designs and identification of relevant parameters in order to achieve stimulation of transfected cells.

3pBA9. A feasibility study on using external microphone measurements for reconstructing *in vivo* interskeletal forces. Brandon McChristian (Mech. Eng., The Univ. of Alabama, Tuscaloosa, AL 35487, bcmcchristian@crimson.ua.edu) and W. S. Shepard (Mech. Eng., The Univ. of Alabama, Tuscaloosa, AL)

While there is much interest in the measurement of forces within the skeleton during human movement, the need for *in vivo* sensors makes measuring these forces challenging. This study investigates the feasibility of using a non-invasive external measurement technique utilizing microphones on the surface of human skin to enable characterization of impulse forces propagating through bone structures. The challenge lies in the ability to measure pressure waves that propagate through complex human tissue. A simplified anatomical model of a human leg is examined with a hollow metal bar representative of a simplified bone structure and a coupled homogeneous ballistic gel to represent the muscular and integumentary systems. In the experimental tests, an impulse force was applied to the bar with an impact hammer, and surface-mounted microphones were used to measure the signal generated on the surface of the ballistics gel. Feasibility of this measurement technique for inferring forces will be determined by examining the quality of the measured acoustic signal resulting from coupled vibrations of the bar and gel structure. An objective of the study is to ensure that the measured acoustic signal has sufficient quality to be useable in reconstructing the impulse forces acting on bones via inverse methods.

3pBA10. Using singular value distribution (SVD) of backscattered impulse response matrix to differentiate between low and high porosity in cortical bone. Omid Yousefian (North Carolina State Univ., 911 Oval Dr., Raleigh, NC 27606, oyousef@ncsu.edu), Yasamin Karbalaeisadegh, and Marie M. Muller (North Carolina State Univ., Raleigh, NC)

The goal of this study is to investigate whether singular value distribution (SVD) of the impulse response matrix can be used to differentiate between healthy and osteoporotic bone using backscattered signals from cortical bone samples *ex-vivo*. Two human cortical bone samples from the tibia were used, with a low (11% porosity, 64 μm average pore diameter, 17 pore/mm^2) and high porosity (30% porosity, 105 μm average pore diameter, 14.73 pore/mm^2). In order to take the effect of randomness into account, for each sample, three measurements at different locations of tibia were performed. The propagation matrix consisting of backscattered signals was collected by measuring the impulse response between each pair of elements of a linear array transducer (7.8 MHz central frequency). The singular value distribution of the propagation matrix in the frequency domain was evaluated for overlapping 0.5 μs time windows in the 6–9 MHz range. After normalizing, the probability density function of the SVD was evaluated and the strongest singular value was compared for low and high porosities. The results show a 20% difference on average for the largest singular value between high and low porosity, showing a potential of this method to differentiate between healthy and osteoporotic bone.

3pBA11. Remote hydrogelation by metal-ligand coordination using high intensity focused ultrasound. Umesh Jonnalagadda (School of Chemical and Biomedical Eng., Nanyang Technolog. Univ., 62 Nanyang Dr., Singapore 637459, Singapore, umeshsj@ntu.edu.sg), Minh Nguyen (Inst. of Chemical and Eng. Sci., A*STAR, Singapore, Singapore), Feifei Li (School of Physical and Mathematical Sci., Nanyang Technol. Univ., Singapore, Singapore), Jim Lee (Inst. of Chemical and Eng. Sci., A*STAR, Singapore, Singapore), Xu Liu, Atsushi Goto, and James J. Kwan (School of Chemical and Biomedical Eng., Nanyang Technol. Univ., Singapore, Singapore)

Smart materials have garnered interest due to their potential and versatility in a wide range of applications from tissue engineering, drug delivery, and cosmetics. It should be appreciated that while ultrasound has been utilized as a stimulus for smart materials, including organogels, its application has primarily been focussed below 100 kHz and for chemical scission. To investigate the potential of high intensity focussed ultrasound (HIFU) for gel formation, we synthesized a water-soluble comb-like polymethacrylate copolymer with randomly distributed tridentate ligands. We utilized reversible metal-ion non-covalent interactions in the design of our polymeric system to bind monomeric units together into temporary crosslinking networks or polymeric aggregates. To minimize the immediate formation of the metallogel, other functional groups within the polymer were involved in competitive reversible binding processes with the solvated metal ions. We show that HIFU triggers hydro-metallogelation by breaking the weaker labile interactions that in turn may allow for more favourable metal ion-ligand chelation, resulting in the formation of a gel network. Considering the established targeting potential of HIFU, our results here open a framework for targeted *in situ* triggered acoustic hydro-metallogelation for biomedical and industrial applications.

3pBA12. Impact of scatterer motion on the correlation of pulse-echo ultrasound signals. Dongwoon Hyun (Radiology, Stanford Univ., 3155 Porter Dr., Palo Alto, CA 94304, dongwoon.hyun@stanford.edu) and Jeremy J. Dahl (Radiology, Stanford Univ., Palo Alto, CA)

The correlation of two pulse-echo signals is an essential component of numerous medical ultrasound imaging methods, including blood flow imaging, elastography, and phase aberration correction. Here, we use an impulsional formulism to derive an expression for the correlation of two arbitrary pulse-echo signals from diffuse scatterers undergoing a bulk uniform motion. The expression is shown to coincide with prior derivations for stationary scatterers. The expression was evaluated numerically and compared against matched simulations of pulse-echo speckle signals. For delay estimation, transverse scatterer motion was found to increase estimator jitter, whereas axial motion increased jitter only for broadband apertures. Delay estimation with synthetic transmit apertures was robust to scatterer motion despite being acquired over a distributed period of time. Receive aperture coherence measurements with synthetic transmit apertures were unaffected by motions of 100 μm laterally and 10 μm transversely or smaller, but were decorrelated by larger motions. In each examined application, close agreement was observed between the theoretical predictions and the simulation measurements, with larger errors observed near the aperture where the Fresnel approximation was less valid. The presented theory provides a motion-aware characterization of the correlation of arbitrary pulse-echo ultrasound signals.

3pBA13. Assessment of feline chronic kidney disease (CKD) using ultrasound Diffusion Constant. Yasamin Karbalaeisadegh (Mech. and Aerosp. Eng., North Carolina State Univ., 2704 Brigadoon Dr. Apt A, Raleigh, NC 27606, ykarbal@ncsu.edu), Ryan Appleby, Gabriela Seiler (College of Veterinary Medicine, North Carolina State Univ., Raleigh, NC), and Marie M. Muller (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC)

The indicator of kidney function, glomerular filtration rate is time and resource intensive, and blood and urine tests are insensitive. Hence, there is a need for non-invasive and sensitive methods to identify chronic kidney disease (CKD). Contrast enhanced ultrasound was used to measure the diffusion constant which is expected to change with changes in vascularization and microbubble population. Data was obtained on 43 cats (23 controls, 10 stage 1, 10 stages 2–4) after injection of 0.05 ml of Definity contrast agent. The inter-element response matrix was acquired by transmitting 5 MHz pulses with individual elements of a transducer array (L7-4) using a Verasonics Vantage system. The incoherent contribution (I_{inc}) to the backscattered intensity was extracted. The diffusion constant (D) was measured using the spatial growth of I_{inc} over time. A significant decrease ($p = 0.0021$) in D was observed in cats with stages 2–4 chronic kidney disease (mean D: 1.04 mm^2/μs), compared to control (mean D: 1.69 mm^2/μs), and stage 1 CKD cats (mean D: 1.77 mm^2/μs). Smaller kidneys and presence of structural abnormalities could result in a denser vasculature, leading to more multiple scattering by the microbubbles and consequently lower D values for stage 2-4 CKD cases.

3pBA14. Acoustic droplet vaporization embedded in 3-D printed gelatin methacrylate hydrogels. Jenna Osborn (Mech. and Aerosp. Eng., George Washington Univ., Ste. 3000, 800 22nd St. NW, Washington, DC 20052, jennakosborn@gwu.edu), Lijie Grace Zhang (Mech. and Aerosp. Eng., George Washington Univ., Washington, DC, DC), and Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., Washington, DC)

Gelatin Methacrylate (GelMA) is a popular hydrogel used for tissue engineering applications due to its biocompatibility, tunable mechanical properties, and rapid reproducibility. With its ability to be crosslinked with ultraviolet light exposure, GelMA can be 3-D printed into a variety of different shapes and geometries for customization for patient needs. We embedded phase-shift perfluorocarbon nanodroplets within the GelMA resin prior to 3-D printing. The droplets were then vaporized by external acoustic excitations determining the acoustic droplet vaporization (ADV) and inertial cavitation (IC) thresholds of the embedded droplets. The ADV of embedded droplets in a tissue engineering scaffolds can be used for triggered release of droplet contents. Varying the size and liquid core composition of the droplets as well as the mechanical properties of GelMA, one can change the ADV and IC characteristics of the embedded droplets. We will present the results of our experiments and discuss their implications on the potential use of this technology in tissue engineering.

3pBA15. Low intensity, continuous wave focused ultrasound reversibly depresses heart rate in anesthetized mice. Ethan Bendau (Biomedical Eng., Columbia Univ., 630 W 168th St., P&S 19-418, New York, NY 10032, ethan.bendau@columbia.edu), Christian Aurup, Hermes Kamimura, and Elisa Konofagou (Biomedical Eng., Columbia Univ., New York, NY)

Focused ultrasound (FUS) can noninvasively and reversibly modulate brain activity. Observed motor activity or behavioral changes are commonly used to infer neuromodulatory effects. Changes in autonomic nervous

system regulation such as heart and respiratory function can potentially be used for studying the effects of FUS neuromodulation that do not produce outwardly observable motor or behavioral activity. In this study, low pressure (850 kPa), long duration (120 s) continuous-wave (CW) FUS at 2 MHz was targeted such that the focus spanned the visual cortex, hippocampus, and thalamus. These parameters were shown to reversibly depress heart rate in *in vivo* mice anesthetized with sodium pentobarbital by an average of 8.1 $\pm$ 3.7%. Following stimulus offset, heart rate either increased or varied in trend. Higher pressure (2 MPa), short duration (0.3 s) pulses at the same frequency resulted in either small, transient increases in heart rate or no change. Thermocouple measurements show the peak temperature elevation in the dura to be approximately 2 °C. Thermal simulations predict a peak temperature increase in the brain of 2.3 °C. These results indicate that CW FUS can alter autonomic nervous system regulation through very low heating and show that cardiorespiratory responses can be altered as a result of FUS neuromodulation.

WEDNESDAY AFTERNOON, 4 DECEMBER 2019 STUART, 1:00 P.M. TO 2:35 P.M.

Session 3pEA

Engineering Acoustics, Physical Acoustics, and Structural Acoustics and Vibration: Acoustic Holography and Visualization of Sound: Methods and Applications

Michael V. Scanlon, Chair
RDRL-SES-P, Army Research Laboratory, 2800 Powder Mill Road, Adelphi, Maryland 20783

Invited Papers

1:00

3pEA1. Acoustic excitation and detection of guided waves in a submerged aluminum plate using an angular spectrum approach. Kyle S. Spratt (Appl. Res. Labs., The Univ. of Texas at Austin, The University of Texas at Austin, Austin, TX 78713, sprattkyle@gmail.com), Benjamin C. Treweek (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Michael R. Haberman, and Mark F. Hamilton (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

An angular spectrum approach can be used to obtain high-resolution measurements of the acoustic plane wave reflection and transmission coefficients corresponding to a given planar sample of material. Such an approach involves exciting the sample with a compact broadband source located in the near field and then scanning the resulting reflected or transmitted field on a plane parallel to the sample surface. Fourier transforms in both space and time then yield a description of the acoustic field in terms of its angular spectrum, or expansion into time-harmonic plane waves propagating throughout a range of frequencies and angles. To demonstrate this method, measurements made on a thick aluminum plate submerged in water are shown. In this case the reflection and transmission coefficients are intricately structured, with individual features corresponding to different symmetric and antisymmetric guided wave modes propagating within the plate. The coincidence effect, by which the plate is nearly transparent acoustically at just those frequency-angle pairs that excite guided waves, is clearly seen in the measurements. Measurements made on other materials, as well as possible extensions to this method, will also be discussed. [Research supported by ONR.]

1:25

3pEA2. Determining structural acoustic properties from noise-based holographic measurements. Simone Sternini (Scripps Inst. of Oceanogr., UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093, sisternini@ucsd.edu), Jit Sarkar (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA), Sandrine Rakotonarivo (Aix Marseille Univ, CNRS, Centrale Marseille, Marseille, France), Alexis Bottero (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA), Earl G. Williams (Code 7106, Naval Res. Lab., Washington, DC, DC), Jeffery D. Tippmann (Intelligence and Space Res. Div., Los Alamos National Lab., Los Alamos, NM), and William A. Kuperman (Scripps Inst. of Oceanogr., UC San Diego, La Jolla, CA)

A facility has been constructed to determine the scattering properties of an arbitrary object in an external medium by measuring its response to a noise field in a laboratory. The measurements are made using a holographic array of a double layer of MEMS microphones placed in the near field of the object; the array is embedded in a 3-D printed scaffold structure which surrounds the object. Cross correlating the MEMS acoustic data together with holographic signal processing yields the structural impedance matrix which together with additional straightforward computation provides the scattering properties of the object when placed (loaded) in an external medium [*J. Acoust. Soc. Am.* **134**, (6) (2013); *J. Acoust. Soc. Am.* **142**(1) (2017)]. The theory underlying the experimental procedure is explained together with results from measurements of a capped cylindrical shell that are also compared to finite element calculations. [Work partially supported by the Office of Naval Research.]

Contributed Papers

1:50

3pEA3. High-resolution remote acoustic localization of damage in vibrating plates. Tyler J. Flynn (Mech. Eng., Univ. of Michigan, Ann Arbor, 1231 Beal Ave., Ann Arbor, MI 48109, tjayflyn@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, Ann Arbor, MI)

Two major challenges exist for the remote acoustic localization of damage in vibrating structures. First, the presence of damage typically results in small acoustic changes that may become "drowned out" by the majority of the sound which remains unchanged. To mitigate this, a high-resolution beamformer can be used to better resolve the acoustic sources, including the small changes (i.e., damage). However, this elicits the second challenge; many popular high-resolution beamformers exhibit poor performance in the presence of coherent sources—exactly the type of sources provided by extended, vibrating structures like a plate. To address both challenges, the Spectral Estimation Method with Coherent Background Removal (SEMCBR) is introduced, enabling the localization of small acoustic changes in a distribution of coherent sources. SEMCBR is an extension of the existing Spectral Estimation Method with Additive Noise (SEMWAN), which has previously been applied to quantify incoherent aeroacoustic sources, combined with an additional subarray averaging step to suppress coherent cross-terms. Experimental results using SEMCBR are provided for the localization of cuts, fastener faults, and delamination in 30-×-30-×-0.16 cm plates base-excited from 100–6000 Hz, using an 8-×-8 array of remote receivers. [Sponsored by NAVSEA through the NEEC and by the US DoD through an NDSEG Fellowship.]

2:05

3pEA4. Optimization of linear and planar array geometries for unmanned underwater vehicle acoustic imaging techniques. Blaine M. Harker (Sensors and Sonars Dept., Naval Undersea Warfare Ctr. Div. Newport, 1176 Howell St., Newport, RI 02841, blaineharker@gmail.com) and Daniel P. Hopper (Sensors and Sonars Dept., Naval Undersea Warfare Ctr. Div. Newport, Newport, RI)

Unmanned underwater vehicles (UUVs) are becoming more prevalent in industrial and scientific applications. A wide range of designs is available and many platforms are highly configurable, resulting in an ever-changing acoustic profile. Changes in the UUV's noise may affect onboard sensors and potentially disturb the environment. An acoustic source imaging process is under development to provide a detailed analysis of the UUV noise in a rapid-turnaround environment. Measurements from a hydrophone array in the geometric near field of a UUV may be input to an acoustic inverse method [e.g., generalized inverse beamforming (GINV) or statistically optimized near-field acoustical holography (SONAH)] to spatially separate noise components and ascertain their levels. However, source resolution and accuracy are highly dependent on array design. Results of a recent numerical case study are presented towards the optimization of linear and planar array geometries. The array geometry is iterated and tuned based on the resultant beamwidth, sidelobe levels, and source localization accuracy for a range of octave frequency bands and source locations. The optimization results provide insight into the critical design factors for acoustical imaging techniques, as well as a suitable array geometry for improved UUV acoustic imaging.

2:20

3pEA5. Facial proximity detection based on inaudible noise for mobile authentication. Yegor Sinelnikov (Zebra USA, 126 Liberty Ave., Port Jefferson, NY 11777, ysinelnikov@yahoo.com), Russell Calvarese (Zebra USA, Holtsville, NY), Larry Hawker (Zebra CA, Mississauga, ON, Canada), William Sakoda, and Andrew Boyden (Zebra USA, Holtsville, NY)

Enterprise-class mobile devices find widespread use in the factories, warehouses, hospitals. A complement of audio communication features enables workers to maximize their productivity and improve customer service. We demonstrate that acoustic interaction of inaudible noise delivered by a mobile device with nearby physical objects has a potential for real-time proximity detection and facial recognition. The mobile device was mounted near the head and torso simulator inside the listening room. The device receiver produced low level inaudible equalized above 15 kHz noise signal, which was recorded by one of the device's microphone. The received signal was measured at different device orientations and distances relative to the simulator's artificial ear. The measurements revealed good correlation between received signal power spectral density and distance to artificial ear. Positive correlation with artificial ear shape was also observed. The results have applicability in the field of mobile communication security and potential for instant biometric authentication.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

James P. Cottingham, Chair
Physics, Coe College, 1220 First Avenue NE, Cedar Rapids, Iowa 52358

Chair's Introduction—1:00

Invited Papers

1:05

3pID1. Overview of efforts within ASA to promote effective communication of science to the public. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, andy.piacsek@cwu.edu) and Laura Kloepper (Biology, St. Mary's College, Notre Dame, IN)

As the fruits of science and technology have permeated nearly every facet of modern life, and public policy decisions increasingly rely on knowledge created by the scientific process, there is growing concern that a deficiency of scientific literacy contributes to poor decisions in both private and public spheres. The ASA, consistent with its mission to generate, disseminate, and promote the knowledge and practical applications of acoustics, is working to address this deficiency by facilitating communication between its members and journalists. The Public Relations Committee, together with Education in Acoustics and Student Council, have organized four special sessions since 2011 on the subject of effective communication of science to non-scientists. At the spring 2018 meeting in Minneapolis, Student Council organized a science communication workshop aimed at students and early career members. At the current meeting in San Diego, the Monday Tutorial Session is an active participation workshop, led by media professionals, designed to boost the skills and comfort level of ASA members in effectively communicating technical concepts, as well as the underlying scientific process, to members of the media. A summary of these efforts, and their significance for the Society, will be presented.

1:25

3pID2. Communication between native and non-native speakers. Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, mbaesebe@uoregon.edu)

For decades, research in speech communication has ask how an individual perceives and produces speech. However, most prior research has used an idealized, monolingual speaker-listener as its model, which is not the norm in our globalizing world. Real-world situation of communication between native and non-native speakers can result in challenges for successful communication. However, the sources of those miscommunications are not well-understood. While many studies have directly examined how a learner acquires a second language, only in the last decade has research expanded rapidly on how a native listener might improve their ability to understand non-native speech. An understanding of the communication between native and non-native speakers is required to truly understand the processes that underlying speech communication, broadly. In this talk, I address issues of communication between native and non-native speakers in their capacities as speakers and listeners. Specifically, I describe the current state of knowledge about how non-native speakers understand and produce speech in their second (or third) language, how native speakers understand non-native speech, and how both parties can improve their abilities at these tasks. This line of inquiry has implications for both practical goals of improving communication outcomes and theoretical understanding of speech perception and production.

1:45

3pID3. Hot topics in a warming ocean: How acoustical oceanography can help monitor climate change. Gabriel R. Venegas (Appl. Res. Labs., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78712-1591, gvenegas@utexas.edu)

Global CO_2 concentrations are higher than they have been over the last 800 000 years (Luthi *et al.*, 2008) and continue to rise, leading to what is thought to be the fastest increase in ocean acidity in the last 60 million years (Turley and Gattuso, 2012). The increase in greenhouse gasses in the atmosphere warms the planet and its oceans, which decreases the solubility of CO_2 in the ocean and instigates a positive feedback loop. As a result, glaciers and polar icecaps are melting, extreme weather patterns are occurring, and the existence of valuable ecosystems that help offset anthropogenic greenhouse gas emissions are being threatened. This talk will focus on past and current developments in acoustical oceanography to help monitor climate change from below the ocean's surface, such as acoustic tomography for measuring ocean temperature; the use of ambient noise and variation in transmission loss to estimate water column properties like ocean acidity or boundary processes like the melting of glaciers and polar icecaps; and acoustic monitoring techniques for underwater methane seeps and carbon storage/fluxes in ocean bottom sediments and seagrass meadows.

Session 3pMU

Musical Acoustics, Signal Processing in Acoustics, and Computational Acoustics: Machine Learning in Musical Acoustics

Bozena Kostek, Cochair
Gdansk University of Technology, Narutowicza 11/12, Gdansk 80-233, Poland

Scott H. Hawley, Cochair
Chemistry & Physics, Belmont University, 1900 Belmont Blvd, Nashville, Tennessee 37212

Chair's Introduction—1:30

Invited Papers

1:35

3pMU1. SPNet: Automated object detection of antinode regions in oscillating steelpan drums. Scott H. Hawley (Dept. of Chemistry & Phys., Belmont Univ., 1900 Belmont Blvd, Nashville, TN 37212, scott.hawley@belmont.edu) and Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, Joliet, IL)

We present a machine learning computer program which applies a neural-network-based object detection algorithm to individual frames of high-speed video of oscillating Caribbean steelpan drums obtained via electronic speckle pattern interferometry (ESPI). This algorithm is trained in a supervised learning context on a dataset of crowd sourced human-annotated images obtained from the Zooniverse Steelpan Vibrations project. The computer code, which we call "SPNet," is subsequently able to annotate new frames of similar images in a manner consistent with the humans' prior work, albeit much more quickly—hundreds of frames per second. The goal of this annotation work is to better understand the dynamical behavior of these drums such as the coupling of oscillations in different parts of the drum surface, by tracking the development of sympathetic vibration modes and extracting their relevant physics. We present details of the algorithm, performance metrics, and some preliminary physics results.

2:00

3pMU2. Learning the nuance of musical instrument acoustics. Chris Donahue (Comput. Sci., Stanford, 9344 Redwood Dr. Apt. D, Palo Alto, CA 94305, cdonahue@ucsd.edu)

While acoustic modeling successfully captures the broad strokes of how musical instruments produce sound, it falls short of capturing the more nuanced attributes required to synthesize entirely convincing reproductions. For example, there are innumerable (effectively random) factors that affect the waveform produced by a single violinist performing the same musical gesture multiple times. Using machine learning, we can implicitly learn a distribution of these factors by modeling a collection of instrument waveforms, which could potentially lead to more convincing synthesis. In this talk, I will discuss our work on using generative adversarial networks, an unsupervised machine learning technique, to synthesize instrument waveforms. I will also speculate about how similar strategies might be paired with existing acoustic models to produce digital instruments which are indistinguishable from their real counterparts.

2:25

3pMU3. Music information retrieval—The impact of technology, crowdsourcing, big data, and the cloud in art. Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

The exponential growth of computer processing power, cloud data storage, and crowdsourcing model of gathering data bring new possibilities to Music Information Retrieval (MIR) field. MIR is no longer music content retrieval only; the area also comprises the discovery of expressing feelings and emotions contained in music, incorporating other than hearing modalities for helping this issue, users' profiling, merging music with social media and qualitative recommendations in music services. Moreover, 5G telecommunications networks, characterized by "near-instant and everything in the vicinity talks with one another," with exponentially faster download and upload speeds, may change the existing models and create a new age of interconnectedness. This paper aims at showing some of the already highly exploited technologies and crowdsourcing models applied to music processing. Several studies are discussed in details, such as, e.g., deep learning applied to music, a way to generate an expanded training sets using 2-D data such spectrograms, mel-cepstrograms, chromagrams, and waveform-based representations of the signal instead of feature vectors in machine learning, allowing to retain all nuances related musical articulation in the signal. Also, a discussion is to be outlined, expanding the issue of the impact of these new technologies on the artistic and aesthetic values of music.

2:50

3pMU4. Convolutional neural network for chordophones recognition. Nicholas J. Dutz (The Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC 20064, 56dutz@cua.edu), Nick Gangemi (Mech. Eng., US Naval Res. Lab., Washington, DC), Andrew Cunningham, Michael Kvartunas, Amelia Vignola, Joseph F. Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Hannah Kurdila (US Food and Drug Administration, Silver Spring, MD)

A student project at Catholic University related to string instruments compared the radiated sound from a guitar, a banjo and 10 ukuleles using a convolutional neural network, CNN. The objective was to determine if the CNN could be trained to distinguish between the instruments and classify them. The guitar, banjo and one of the ukuleles were factory built and the remaining 9 ukuleles where built by the team, four from kits and five from scratch. There were at least 100 single note strikes recorded for each instrument and each instrument was recorded in three different acoustic environments, anechoic, highly reverberant and moderate reverberation. This presentation will discuss the construction process used to build the ukuleles and test method used for recording. Additionally, data comparing the scratch build instruments to comparably sized factory built ukuleles will be presented to show that the CUA built instruments were appropriate for evaluation. Finally, the presentation will show the network's predictions for instrument type and recording environment.

3:05

3pMU5. PhonoNet: Multi-stage deep learning for raga preservation in hindustani classical music. Sauhaarda Chowdhuri (Westview High School, 8376 Entreken Way, San Diego, CA 92129, sauhaarda@resoniq.com)

Hindustani classical music is an ancient improvisational form of music based on ragas, melodic frameworks which have no written representation and are passed down through a fading oral tradition. The proposed system, PhonoNet, aims to preserve Hindustani classical music by providing computational prediction of ragas, so singers can receive live feedback when learning. PhonoNet also creates a visual format for documenting and preserving raga information. First, the system computes the short-term Fourier Transform of the input audio data to form a chromagram representation of the notes being sung. These data are then augmented using a transpositional data augmentation algorithm and split into chunks for use as training inputs for a deep convolutional neural network. The convolutional network's filters are analyzed using a saliency visualization algorithm and modified with a recurrent layer to allow processing of full-length songs. The convolutional system achieves 78.9% validation accuracy for raga prediction on 150 second audio chunks. The joint raga prediction system achieves a new state-of-the-art 98.9% accuracy for raga prediction on full-length songs. Future work can extend the proposed hierarchical system to other tasks with long temporal sequences and extend the data augmentation and visualization algorithms to different applications of audio processing.

3:20

3pMU6. Discovering rule-based learning systems for the purpose of music analysis. Grazina Korvel (Inst. of Data Sci. and Digital Technologies, Vilnius Univ., Vilnius, Lithuania) and Bozena Kostek (Audio Acoust. Lab., Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, bokostek@audioakustyka.org)

Music analysis and processing aims at understanding information retrieved from music (Music Information Retrieval). For the purpose of music data mining, machine learning (ML) methods or statistical approach are employed. Their primary task is recognition of musical instrument sounds, music genre or emotion contained in music, identification of audio, assessment of audio content, etc. In terms of computational approach, music databases contain imprecise, vague and indiscernible data objects. Moreover, most of the machine learning algorithms outcomes are given as a black-box result. Also, underfitting or overfitting may occur, meaning that either the model description is not complex enough or the test set is too small or not sufficiently representative. Thus the goal is to generalize the model. To overcome some of these problems, rule-based systems may be used, e.g., based on rough set theory that shows the outcome in the form of rules interconnecting features retrieved from music. Thus, first, principles of rule-based classifiers and particularly rough sets (RS) are presented, showing their usability in the music domain. A potential of the rough set-based approach was shown in the context of music genre recognition. The results were analyzed in terms of the recognition rate and computation time efficiency.

Session 3pNS

Noise, Psychological and Physiological Acoustics, and ASA Committee on Standards: and Development of New Sounds for Electric Vehicles

Brigitte Schulte-Fortkamp, Cochair
Institute of Fluid Mechanics and Engineering Acoustics, TU Berlin, Einsteinufer 25, Berlin 10623, Germany

Klaus Genuit, Cochair
HEAD Acoustics GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany

Chair's Introduction—1:00

Invited Papers

1:05

3pNS1. EMobility – The silent revolution!? Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de)

Electric vehicle become more and more popular worldwide. A strong increase of new registered electric vehicles is expected. The interior and exterior sounds of electric vehicles will become an important aspect for sound engineering considering the psychoacoustics: what sounds do customers prefer, what kind of sounds are important for customers, how to consider the typical acoustical orientation known from combustion noises, should it be possible, that a customer can actively/individually influence the character of the vehicle interior noise, which new sounds arise and must be avoided or emphasized, how to design vehicle exterior noise (e.g., pedestrian protection)? Based on the ISO 12913 Soundscape standard the following topics will be discussed: concepts for the development of sounds, applying jury testing to predict human reactions towards the noise or sound? of new vehicle types, application of psychoacoustics analysis and simulation of warning signals in complex traffic scenarios to check their applicability and their annoyance.

1:25

3pNS2. Electric car sound design—The paradox of almost unlimited freedom. M. Ercan Altinsoy (Chair of Acoust. and Haptics, Technische Universitaet Dresden, Helmholtzstr. 18, Dresden 01062, Germany, ercan.altinsoy@tu-dresden.de)

Electric and hybrid cars are relatively silent at low speeds. Therefore sound designers have an almost unlimited freedom regarding the electric and hybrid car indoor sound design. The extensive experiences with Active Sound Design (ASD) increases our abilities. However at least in a very near future, customer expectations are based on our long term experience with vehicle sounds with conventional engines. This point leads the designers to design new car sounds which have extensive similarities with the combustion engine car sounds. Some car manufacturers try to break this trend designing extraordinary sounds. But most of these ideas are based on previous film sound design examples and therefore the originality is limited. At the same time, the reaction of the customers to such kind of sounds is not very positive. Another important point is that there are already regulations, which define the requirements for the Acoustic Vehicle Alerting Systems (AVAS), for the exterior sound of electric and hybrid cars. In some cases, the sound of AVAS interacts with the interior sound. In this talk, the various aspects of future electric car sound design will be discussed according to our previous investigations.

Contributed Paper

1:45

3pNS3. Design and realisation of a directional electric vehicle warning sound system. Nikolaos Kournoutos (Inst. of Sound and Vib. Res., Univ. of Southampton, University Rd., B13/3049, Southampton, Hampshire SO17 1BJ, United Kingdom, nk1y17@soton.ac.uk) and Jordan Cheer (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, Hampshire, United Kingdom)

The mandatory use of artificial warning sounds by electric vehicles at low speeds has generated concern over the resulting increase in urban noise levels. Systems have been proposed and tested that limit the vehicle's environmental noise contributions by directing the radiated sound solely towards vulnerable road users. However, such solutions are often too costly to manufacture and maintain as they tend to feature loudspeaker arrays installed on the vehicle and exposed to the environment. This paper proposes a directional warning sound system for electric vehicles which uses an array of structural actuators attached to a panel of the vehicle's body. The required warning sound is radiated by the vibrating panel and its directivity is controlled by adjusting the amplitude and phase of the signals driving the structural actuators. An analytical model is used to perform a simulation based parametric study of the proposed system, and this is then experimentally validated through measurements of a prototype using a flat metal panel. The effectiveness of the actuator array installed in an actual passenger car is then evaluated in a semi-anechoic environment. Directivity measurements are performed for different target zones using the array in a number of configurations on the vehicle's body.

Invited Papers

2:00

3pNS4. EV sound hacking Part 1: Analyzing the Hyundai/Kia VESS. Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

Current Hyundai and Kia EV models are delivered with the Virtual Engine Sound System (VESS), which produces a speed-dependent alert sound akin to a cartoon spaceship. This paper details the process of reverse engineering the VESS and associated CAN bus for the purpose of operating the device while disconnected from the car, and results of acoustical measurements. These include sound level, directivity, and speed-dependent timbre.

2:20

3pNS5. EV sound hacking Part 2: Creating an alternative soundtrack. Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

The Hyundai VESS (detailed in Part 1) is controlled by one of the vehicle's CAN (controller area network) buses. Messages containing speed, braking, power, and dozens of other parameters are continuously streamed to the bus and can be decoded by a simple microcontroller or single-board computer. The author, an avid railroad enthusiast, sought to replace the standard warning sounds in his Hyundai Kona EV with those of a locomotive. A microcontroller is used to translate CAN messages to DCC, a standard model railroad control protocol, to facilitate the use of an off-the-shelf model railroad sound module. This allows for speed-dependent locomotive sounds to be generated automatically by the electric vehicle.

2:40–3:00
Panel Discussion

WEDNESDAY AFTERNOON, 4 DECEMBER 2019 — WILDER, 1:00 P.M. TO 3:00 P.M.

Session 3pPA

Physical Acoustics: General Topics in Physical Acoustics III

Michael B. Muhlestein, Chair
Cold Regions Research and Engineering Laboratory, U.S. Army Engineering Research and Development Center, 72 Lyme Rd., Hanover, New Hampshire 03777

Contributed Papers

1:00

3pPA1. Mimicking a perfectly matched layer with a porous medium. 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) and Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., U.S. Army Engineer Res. and Development Ctr., Hanover, NH)

The possibility of designing a porous medium that mimics a perfectly matched layer (PML) is discussed. In numerical methods, a PML provides essentially zero reflection and rapid attenuation in space. As a starting point to designing such a medium, the relaxation-model equations (10.54) and (10.55) from Ostashev and Wilson [*Acoustics in Moving Inhomogeneous Media* (CRC Press, Boca Raton, 2016)] are employed. In general, it does not appear to select the parameters to mimic a PML across a broad range of frequencies. For relatively high frequencies, however, it is shown that by setting the ratio of the vorticity and entropy shape factors to $\sqrt{N_{pr}}/(\gamma\text{-}1)$, where N_{pr} is the Prandtl number and γ the ratio of specific heats, the impedance becomes real-valued while the medium remains attenuative. The reflected power is small provided that the porosity is close to one. It is unclear whether such media are physically realizable. Possible approaches to realization are discussed based on varying the pore shape, parallel configurations of pores with differing cross sections, and metamaterials.

1:15

3pPA2. Calculation of wave dispersion curves in multilayered composites using semi-analytical finite element method. Prachee Priyadarshinee (Mech. Eng., National Univ. of Singapore, #21-256B, 38 College Ave. East, Singapore 138601, Singapore, a0135595@u.nus.edu)

The enormous potential of Guided waves have a significant use for non-destructive evaluation(NDE) and structural health monitoring(SHM). It is common to use different solution approaches to predict dispersion curves depending on the material type, isotropic, transversely isotropic, or orthotropic, of the medium in which the wave propagates, shape of the waveguide, etc. Inability to model waveguides of arbitrary cross-section by existing matrix methods has led to development of Semi-Analytical Finite Element(-SAFE) Method. In this study, rapid and accurate prediction of the propagation characteristics for guided waves in a wide range of structures has been performed by implementation of SAFE method. Damped and undamped waveguides of different shapes can be modelled. Wave propagation characteristics in both isotropic and anisotropic composite structures can also be studied. To benchmark the solution, a case study using Transfer Matrix Method (TMM) for composite plate has been evaluated. It was found that the element discretizaton plays an important role in accurate prediction of modes in SAFE method, neglecting which leads to loss of higher order modes.

1:30

3pPA3. Dispersion curves identification and traveling wave control in a weakly coupled impedance tube. Yoav Vered (Mech. Eng., Technion - Israel Inst. of Technol., Dynam. Lab. Mech. Eng. Faculty, Technion, Haifa 3200003, Israel, syoavv@gmail.com), Ran Gabai, and Izhak Bucher (Mech. Eng., Technion - Israel Inst. of Technol., Haifa, Israel)

Impedance tubes are commonly used for non-destructive measurement method of the acoustic properties of materials. Most physical models neglect the elastic nature of the tube resulting in a single mode propagation up to the cut-off frequency and are thus unable to accurately predict the propagation patterns in the case of liquid-filled tubes. An understating of the dispersion pattern in the latter case is crucial for the success of the acoustic properties estimation procedure. Furthermore, a model-based control can be implemented on the basis of the dispersion relation which is able to enhance the existing methods. This work presents a case study of an air-filled impedance tube. A novel, two-actuators phase-perturbations technique was developed to identify the tube dispersion curves and to compare to analytical models. The accurate identification of dispersion curves was utilized in a successful effort to control a single mode traveling wave ratio by employing a feedforward control algorithm. The success of the case study in the case of a weak acoustic coupling mechanism such as the air-filled impedance tube emphasizes the capabilities when considering a stronger acoustic coupling such as in the liquid-filled impedance tube.

1:45

3pPA4. The physics of knocking over LEGO minifigures with time reversal focused vibrations. Lucas A. Barnes (Dept. of Phys. & Astronomy, Brigham Young Univ., BYU, N283 ESC, Provo, UT 84602, lucas.barnes2@gmail.com), Aaron C. Brown (Mech. Eng., Brigham Young Univ., Provo, UT), and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Time reversal (TR) is a method of focusing wave energy at a single point in space. This project optimizes a TR demonstration designed to knock over one selected LEGO minifigure among other minifigures by focusing vibrations within an aluminum plate at the target minifigure. The goal is to achieve high repeatability of the demo along with making it possible to do at a lower cost. By comparing the motion of the minifigure and the plate directly beneath its feet, it was determined that a major factor inhibiting the repeatability of the demo was that smaller vibrations leading up to the focal event caused the minifigure to bounce repeatedly, losing contact with the plate and ending up in the air during the main focal event intended to knock over the minifigure. This project explores the effects of amplitude, frequency, TR technique, plate size, and vibration sensor type on the repeatability of the demonstration. This demonstration illustrates the power of TR focusing and the principles learned by optimizing this demonstration can be applied to other real-world applications.

2:00

3pPA5. Simulated and experimental demonstrations of the first acoustic hologram enhanced phased arrays for manipulation. Luke Cox (Mech. Eng., Univ. of Bristol, UNDT Group, Queen's Bldg., University Walk, Bristol BS8 1TR, United Kingdom, luke.cox@bristol.ac.uk), Kai Melde (Micro, Nano and Molecular Systems Group, Max Plank Inst. for Intelligent Systems, Stuttgart, BW, Germany), Anthony Croxford (Mech. Eng., Univ. of Bristol, Bristol, United Kingdom), Peer Fischer (Micro, Nano and Molecular Systems Group, Max Plank Inst. for Intelligent Systems, Stuttgart, BW, Germany), and Bruce W. Drinkwater (Mech. Eng., Univ. of Bristol, Bristol, United Kingdom)

When attempting to form a desired static sound field for manipulation, acoustic holograms are generally superior to phased arrays due to their immensely higher resolution. However, they lack the dynamic capabilities of phased arrays. We therefore demonstrate a combination of the two. We produce several holograms for use with a continuously excited 64-element linear phased array. Simulations were used to predict the expected range of performance. We then experimentally demonstrate moving the position of the projected hologram plane via phase delays which tilt the output of the phased array. This creates a much more tightly focused point than the phased array alone, whilst retaining dynamical control. A second hologram allows the complex movement of a "phase surfer" along a phase track. These examples demonstrate that the strengths of both phased arrays and holograms can be combined. These developments open the door for more complex manipulation in the future whilst maintaining a relatively simple electronic set-up, thus reducing the cost whilst increasing the capability.

2:15

3pPA6. Fundamental study on enlarge of a loop-tube-type thermoacoustic system—Measurement of onset temperature and sound field in cross-sectional area change. Shin-ichi Sakamoto (Dept. of Electron. Systems Eng., Univ. of Shiga Prefecture, 2500, Hassaka-cho, Hikone, Shiga 522-8533, Japan, sakamoto.s@e.usp.ac.jp), Kenshiro Inui, and Hidekazu Katsuki (Dept. of Electron. Systems Eng., Univ. of Shiga Prefecture, Hikone, Japan)

We have worked to improve the output of the thermoacoustic system to expand the range of system applications. The onset temperature of a loop-tube-type thermoacoustic system was experimentally examined under changes to cross-sectional area in a loop tube. The total length of the system was not varied. This cross-sectional area, i.e., inner diameter was set to values of 24.2, 42.6, and 95.6 mm; onset temperatures at these diameters were found to be 623, 329, and 121 °C, respectively. The sound field in the system was measured to investigate the reason why the onset temperature varies depending on the inner diameter. The sound pressure distribution, particle velocity distribution, sound intensity distribution, and phase difference distribution of sound pressure and particle velocity in the system required for evaluation of the thermoacoustic system were measured. It is confirmed that the sound field is different depending on the inner diameter. It is found that the oscillation frequency is different depending on the inner diameter. The oscillation frequency of system with an inner diameter of 24.2 mm shows two-wavelength resonance. The oscillation frequencies of the other system showed one-wavelength resonance. In addition, dissipation is confirmed to increase as the inner diameter decreased.

2:30

3pPA7. Near shore atmospheric acoustic transmission loss: Numerical predictions. Diego Turo (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., N.E., Washington DC, DC 20064, diegoturo@gmail.com), Teresa J. Ryan (Dept. of Eng., East Carolina Univ., Greenville, NC), Andrea Vecchiotti, John Judge, and Joseph F. Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This work presents advances in the development of a numerical model of atmospheric acoustic propagation designed for a littoral or riverine environment with a near-shore acoustic source and on-shore receivers. A parallel experimental effort uses a measurement system that records concurrent atmospheric and acoustic transmission loss data. The numerical model uses a generalized terrain parabolic equation method to account for sea surface roughness and ground topography along the propagation path. The model accounts for vertical temperature and wind speed profiles as well as surface impedance at the shore. For the comparison with experimental data, wind speed is used to predict the sea state. The propagation range consists of three segments: over water, over swash and sandy shore, and over complex dune with highly varied vegetation of up to 2 m height. Surface impedance estimates for the sandy shore are made based on grain size distribution and impedance tube measurements of excised samples. Existing models for surface impedance of vegetation were used for the dune portion of the propagation range.

2:45

3pPA8. Near shore atmospheric acoustic transmission loss: Method and measurements. Teresa J. Ryan (Dept. of Eng., East Carolina Univ., 248 Slay Hall, Greenville, NC 27858-4353, ryante@ecu.edu), Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC), Julian Quintero-Rivero (Eng., East Carolina Univ., Greenville, NC), John Judge, and Joseph F. Vignola (Mech. Eng., The Catholic Univ. of America, Washington, DC)

A model for moderate to long range outdoor propagation of sound in complex environments must account for a wide variety of parameters. Generally, the characteristics of the source, the space through which the sound propagates, and the receiver must all be considered. This larger project aims to understand atmospheric acoustic transmission loss in the littoral and riverine environments at ranges of up to approximately 3 km. In such an environment, some of the confounding factors in the propagation path include: wind speed and direction, temperature and humidity profiles, absorption, refraction, dispersion, turbulence, diffraction, dynamically changing water surface geometry, shore terrain geometry and surface impedance along the propagation path. A simple pitch-catch configuration with the source over water and the microphone arrays on shore is used to capture transmission loss information, along with concurrent meteorological measurements. In this work, the measurement system is described and preliminary results from measurements at three sites are reported. These measurements inform the concurrent effort to develop and validate an improved model for acoustic transmission loss in these scenarios.

Session 3pSA

Structural Acoustics and Vibration: Topics on Health Monitoring and Non-Destructive Testing

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

Contributed Papers

1:30

3pSA1. Diagnosing wind turbine condition employing a neural network to the analysis of vibroacoustic signals. Andrzej Czyzewski (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl)

It is important from the economic point of view to detect damage early in the wind turbines before failures occur. For this purpose, a monitoring device was built that analyzes both acoustic signals acquired from the built-in non-contact acoustic intensity probe, as well as from the accelerometers, mounted on the internal devices in the nacelle. The signals collected in this way are used for long-term training of the autoencoder neural network. The appropriately trained network automatically detects deviations, signaling them to technical service. The applied methods of analysis of vibroacoustic signals and neural network training are the subject of the presented paper. In addition, the motion magnification of video is used for extracting information on vibrations of the whole wind turbine construction. Finally, spectral analysis is applied for detecting unnatural components presence meaning defects in both: visual and vibroacoustic representations. The process of reduction and construction of a wind turbine model is discussed with a particular emphasis on application to perform extensive tests of the developed methods and algorithms. [The research was subsidized by the Polish National Centre for Research and Development within the project "STEO—System for Technical and Economic Optimization of Distributed Renewable Energy Sources", No. POIR.01.02.00-00-0357/16.]

1:45

3pSA2. Application of no-tachometer time synchronous averaging (TSA) and relative signal strengths to localize gear and bearing faults in a helicopter gearbox. Dan Watson (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, duw428@psu.edu) and Dr. Karl Reichard (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Health and Usage Monitoring Systems (HUMS) installed on helicopters monitor a wide variety of input signals to track system conditions. Improving fault detection on existing platforms precludes installation of additional hardware such as shaft speed sensors and additional accelerometers. The application of no-tachometer TSA and bearing fault harmonic energy techniques can be used to improve HUMS detection ability. In this presentation I will present a case study of gear tooth and bearing fault detection for a helicopter intermediate gearbox (IGB). Gear-mesh frequency demodulation is used to determine angular zero-crossing to synthesize a tachometer signal for gear fault detection, allowing for TSA techniques to readily identify gear tooth faults. The methods presented in "Autonomous Bearing Fault Diagnosis Method based on Envelope Spectrum" (2017 Klausen *et al.*) are then applied to determine the presence of bearing faults inside the IGB. Due to overlapping fault frequencies from multiple bearings a detected fault could be attributed to several bearings, the use of two accelerometer signals to spatially localize faults is developed and presented.

2:00

3pSA3. Experimental analysis of the size and shape of time reversal chaotic cavities for nondestructive evaluation. Paige E. Simpson (Dept. of Phys. & Astronomy, Brigham Young Univ., BYU, N283 ESC, Provo, UT 84602, singingisme18@gmail.com) and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Time reversal may be used as an energy-focusing technique. It is applied in many different ways including, for example, nondestructively evaluating cracks in structures, reconstructing a source event, and providing an optimal carrier signal for communication. In nondestructive evaluation applications, it is often of interest to study small samples or samples that do not lend themselves to the bonding of transducers to their surfaces. A chaotic cavity provides space for the attachment of transducers as well as a more reverberant environment, which is critical to the quality of time reversal focusing. Transducers are attached to the chaotic cavity which is attached to the sample under test. The goal of this research is to explore the dependence of the quality of the time reversal focusing on the size and shape of the chaotic cavity used. An optimal chaotic cavity will produce the largest focusing amplitude, best spatial resolution, and most linear focusing of the time reversed signal. The presentation will provide experimental results for time reversal focusing experiments conducted on various sized and shaped aluminum blocks.

2:15

3pSA4. Structural vibration analysis of Fresco cohesion. Andrew Cunningham (Catholic Univ. of America, Washington, DC), Amelia Vignola (Catholic Univ. of America, The Catholic University of America, Washington, DC, 63vignola@cua.edu), Diego Turo (Catholic Univ. of America, Washington, DC), Francesco Corvaro (Universita Politecnica delle Marche, Ancona, Italy), Teresa J. Ryan (East Carolina Univ., Greenville, NC), Joseph F. Vignola (Catholic Univ. of America, Washington, DC), and Barbara Marchetti (Universita degli Studi e-Campus, Novedrate, Italy)

One approach for non-contact interrogation of mixed-media surfaces, such as fresco, is mapping surface motion induced by acoustic excitation. A scanning laser Doppler vibrometer provides such high spatial resolution surface information which can indicate material inclusions, delaminations, and other subsurface features. This work uses this interrogation method on frescos in the U.S. Capitol Building. These frescos were installed in the mid nineteenth century under the supervision of Italian artist Constantino Brumidi. Brumidi and his team used traditional fresco techniques for their work throughout the U.S. Capitol Building. Specifically, this work focuses on the Senate Reception Room. Such maps of these frescos were completed in 2004. In the intervening years, external events may have changed the fresco integrity. Such events include restoration work performed by the Office of the Architect of the Capitol and the Mw 5.8 earthquake in Mineral, Virginia in 2011. These repeat measurements allow for an assessment of changes due to these events. While these measurements can only locate the region of anomaly, an inverse analysis is required to classify the nature of the anomaly. Therefore, in addition to measurements, numerical finite element simulations are used to classify the type, size, and location of defects.

2:30

3pSA5. Detection of structure anomalies in frescos: A comparison between thermography and laser Doppler vibrometry techniques. Jesse P. Williams (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave. NE, Washington, DC 20064, 44williamsj@cua.edu), Barbara Marchetti (Universita Degli Studi e-Campus, Novedrate, Italy), Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC), Joseph F. Vignola, Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Francesco Corvaro (Industrial Eng. and Mathematical Sci., Universita' Politecnica delle Marche, Ancona, Italy)

This work will investigate two non-destructive methods of fault detection in laminate structures such as frescos. These two techniques are compared on a purpose-built laboratory prototype and on frescos in the Senate Reception Room of the U.S. Capitol Building. Thermography uses an infrared camera to map the surface temperature of an object. This technique can be used to quickly identify abnormally hot or cold regions on a surface. These abnormal regions can indicate structural defects such as fresco delamination, cracks, and inclusions. The other non-destructive technique for defect detection used in this work is laser Doppler vibrometry paired with acoustic excitation of the target surface. Over the same surface area, thermography provides a much quicker result. This work aims to evaluate potential synergy between the technologies by comparing the results of the two scans. A prototype with known defects is used to find commonality between the two methods and to understand the circumstances in which the results diverge. The prototype is a square of approximately half a meter on each side. A seven centimeter thick concrete backing layer is followed by a layer of mortared antique bricks and topped by mortar and plaster to mimic fresco.

2:45

3pSA6. Unbalanced pump detection using modified transfer path analysis. Dhany Arifianto (Dept. of Eng. Phys., Institut Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id) and Nihlatul Falasifah (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia)

Unbalance is the term in a rotating machine that used for a mass distribution which is not rotationally symmetric. Vibration from unbalancing pump affects to other pumps located on the same single base frame due to transmissibility. However, the unbalanced vibration can disturb and give a negative impact to other pumps on the same base frame. Therefore it is important to detect unbalance pump early which transmitted through the base frame because it can save large amounts of money. The aim of this research is evaluating the modification of transfer path analysis method that applied as a detecting pump damage method compare to International Standard. In this research, Modified Transfer Path Analysis (MTPA) is equal to the operational force that found by convolution of the natural frequencies with operational acceleration. The value of natural frequencies are determined using experimental modal analysis and the value of operational acceleration is determined by measuring the acceleration when the vibration source (pump machine) is running on steady condition. ISO 13373-1 was used as the reference for damage detection on the pump system. The results show that MTPA has the same characteristics and in agreement with ISO 13373-1 in detecting unbalance pump condition for machinery condition monitoring.

Session 3pSC

Speech Communication: Neuroscience of Speech Production and Perception and Speech Technology (Poster Session)

Laura Dilley, Chair

Michigan State University, Department of Communicative Sciences and, East Lansing, Michigan 48824

All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Contributed Papers

3pSC1. Assessing the cortical processing of spectrally rotated speech using functional near-infrared spectroscopy. Daniel Friedrichs (Dept. of Speech, Hearing and Phonetic Sci., UCL, Sidgwick Ave., Cambridge CB3 9DA, United Kingdom, daniel.friedrichs@ucl.ac.uk), Kurt Steinmetzger (Dept. of Neurology, Heidelberg Univ. Hospital, Heidelberg, Germany), Andrew Clark (Div. of Psych. and Lang. Sci., UCL, London, United Kingdom), and Stuart Rosen (Dept. of Speech, Hearing and Phonetic Sci., UCL, London, United Kingdom)

Rotated speech (RS) is unintelligible to untrained listeners but contains similar temporal and spectral complexity as ordinary speech. It has been used in many studies to investigate cortical pathways for intelligible speech. Typically, the generation of RS involves the rotation of the spectrum of low-pass filtered speech around a center frequency (e.g., 2 kHz). Due to the inversion, voiced segments of RS contain harmonics that are typically not multiples of the fundamental frequency, and therefore the RS signal is not truly periodic. Here, it was assessed whether cortical responses to such aperiodic RS differ from quasi-periodic RS with preserved harmonic structure. Harmonic RS was constructed by rotating only the spectral envelope while preserving the characteristics of the source, whereas for inharmonic RS both components were rotated. A story listening paradigm was used, in which subjects were presented with 25-s blocks of unprocessed speech, inharmonic RS, and harmonic RS. Functional near-infrared spectroscopy (fNIRS) was used to measure cortical activation based on changes in the concentration of oxygenated and deoxygenated hemoglobin. The results show increased activity in the right posterior superior temporal sulcus (STS) for harmonic RS in comparison with inharmonic RS but not unprocessed speech. This demonstrates that the periodicity of RS should be considered in the interpretation of previous and the planning of future neuroimaging studies. [Wellcome Trust Multi-user Equipment Award and SNSF grant P400PG_180693.]

3pSC2. The perception of frequency modulated sounds in tone and non-tone language speakers: An electroencephalography study. Philip J. Monahan (Linguist, Ctr. for French & Linguist, UTSC, Univ. of Toronto, 1265 Military Trail, Toronto, ON M1C 1A4, Canada, philip.monahan@utoronto.ca), Mayoori Baskarasingham (Michener Inst., Toronto, ON, Canada), and Alejandro Pérez (Linguist, Univ. of Toronto, Toronto, ON, Canada)

Speakers of tone languages acquire expertise in discriminating and identifying frequency modulated auditory signals [Chandrasekaran *et al.*, *Brain Res.*, **1128**, 148–156 (2007); Kaan *et al.*, *Brain Res.* **1148**, 113–122 (2007)], as their native language utilizes such acoustic properties to cue lexical differences. How this expertise influences auditory neurophysiological responses to non-speech stimuli, however, remains poorly understood. We tested adult tone and non-tone language speakers in their automatic brain processing of non-speech frequency modulated (FM) tone chirps. Participants were presented with a series of FM tones in a many-to-one oddball mismatch negativity [MMN; Näätänen and Winkler, *Psychol. Bull.* **125**, 826–856 (1999)] paradigm that varied in whether the modulation was concave or convex in nature and whether the difference between the tone chirp onset and offset frequencies was relatively large or small. The results revealed that the tone group produced a larger MMN than the non-tone group. Moreover, tone language participants produced significantly larger negative deflections in the event-related potential than the non-tone participants in response to both deviant types across a large post-stimulus time-window. Consequently, tone language speakers' expertise in processing frequency cues impacts their neurophysiological responses to non-linguistic stimuli that vary along similar acoustic properties to linguistic stimuli.

3pSC3. Neural commitment alters cue weighting of formant structure in speech perception: A cross-language MEG study. Yang Zhang (Univ. of Minnesota, 164 Pillsbury Dr SE, Minneapolis, MN 55455, zhang470@umn.edu), Patricia Kuhl, Toshiaki Imada (Univ. of Washington, Seattle, WA), and Keita Tanaka (Tokyo Denki Univ., Hatoyama, Saitama, Japan)

This magnetoencephaalography (MEG) study investigated the effects of language experience in cue weighting of formant structure supporting native language neural commitment. The speech stimuli were a grid of /ra-la/ synthetic continuua with systematic variations of the second (F2) and third (F3) formants. The subjects were a group of 10 monolingual adults of American English speakers and a group of 10 adult Japanese speakers. The experimental protocol adopted bi-directional oddball paradigm to examine between-group differences in discriminatory sensitivity at the pre-attentive level to stimulus conditions that featured F2 difference alone, F3 difference alone and F2 + F3 differences. Unlike previous reports, the MEG data showed equivalent amplitudes of mismatch responses for discriminating the /ra-la/ contrast in the F2 + F3 condition between the American and Japanese subjects. In the F2 difference alone condition, Japanese subjects showed enhanced discriminatory sensitivity than Amercian subjects. In the F3 difference alone condition, Japanese subjects showed reduced sensitivity than American subjects. Together, the MEG results provide strong support for an altered pattern of cue weighting that explains the native language perceptual interference in speech categorization.

3pSC4. Event-related cortical potentials occurring prior to speech initiation. Al Yonovitz (Dept. of Communicative Sci. and Disord., The Univ. of Montana, Missoula, MT 59812, al.yonovitz@umontana.edu) and Silas Smith (The Univ. of Montana, Missoula, MT)

Real time brain electrical potentials were obtained in subjects prior to the initiation of speech in a potential clinical paradigm using a single vertex

electrode. The purpose of this research was to establish a real-time event related brain electrical potential system and to compare the results to offline responses. Research in this area in the past has used a great number of facial and lip EMG electrodes. The marking point for determining the pre-event time epoch has been an EMG source. The data are typically acquired offline and later averaged. This research uses a vocal signal as the marking point, and displays in real time the event-related potential. A number of CV stimuli were used. The sample rate (25 600 samples/s) permitted an analysis of both slow negative waves and faster neurogenic signals. Results indicated reliable waveform morphology within and between subjects.

3pSC5. Analysis and modeling for modification of speaking to karaoke-style singing. Soheil Khorram (Univ. of Texas at Dallas, 2400 Waterview Pkwy, Apt. 436, Richardson, TX 75080, khorram.soheil@gmail.com) and John H. Hansen (Univ. of Texas at Dallas, Richardson, TX)

This study considers an analysis of differences in voice modification within the domains of speaking and singing the same text content, resulting in the first attempt in converting speaking to Karaoke-Style singing. To develop this system, we collected a new dataset, UT-Sing dataset, containing more than 23 h of speech from 81 participants across four different languages: English, Farsi, Hindi, and Mandarin. We asked each participant to produce the same text content by first reading and then singing 5 popular songs while listening to the accompanying instrumental music through open-air headphones. This effectively creates a parallel dataset that is suitable for voice modification systems. We first use this dataset to compare different prosodic and spectral characteristics of Karaoke-style singing versus speaking the same text content. We then leverage the knowledge obtained from this comparison to develop a speaking to Karaoke-Style singing modification system. In the training phase of the developed system, we extract acoustic features using the WORLD vocoder; we align the acoustic features through the DTW algorithm. Finally, we train a residual network to model the relationship between the acoustic features. We employ subjective assessments to evaluate the performance of the developed system.

3pSC6. Word recognition and future language skills in 14-month-old infants. Alexis Bosseler (Inst. for Learning & Brain Sci., Univ. of Washington, 1715 NE Columbia Rd. Portage Bay Bldg. Box 357988, Seattle, WA 98195-7988, bosseler@uw.edu), Kambiz Tavabi, Maggie Clarke, Eric Larson (Inst. for Learning & Brain Sci., Univ. of Washington, Seattle, WA), Samu Taulu (Phys., Univ. of Washington, Seattle, WA), and Patricia Kuhl (Inst. for Learning & Brain Sci., Univ. of Washington, Seattle, WA)

Children show remarkable progress in word learning in the second year of life. This language growth coincides with the vocabulary spurt and the development of domain-general cognition, e.g., attention that facilitates perceptual processing. Evidence suggests that language growth depends on increasing processing efficiency and underlying neuronal specialization. We used magnetoencephalography (MEG) to characterize neural activity following familiar and unfamiliar words in 14-month-old infants and related measures of neural activity to prospective measures of vocabulary growth. MEG source modeling revealed a broadly distributed network in bilateral frontal, temporal and parietal cortex that distinguished word classes between 150–900 ms after word onset. Analysis of correlations between outcome measurements and strength of neural activity revealed that in the interval 150–300 ms after word onset, the magnitude of neural activation in the right inferior frontal cortex for familiar words was positively correlated with vocabulary size at 18, 21, 24, 27 and 30 months of age. We argue that increased activity to familiar words may reflect more efficient discriminatory processing by the most skilled language learners who develop a larger vocabulary one year later.

3pSC7. The role of inferior frontal gyrus in rapid conversational turn-taking. Gregg A. Castellucci (Neurosci. Inst., NYU Langone Medical Ctr., 435 E30th St., NYULMC Sci. Bldg., New York, NY 10016, gregg.castellucci@nyulangone.org), Jeremy Greenlee (Neurosurgery, Univ. of Iowa, Iowa City, IA), and Michael Long (Neurosci. Inst., NYU Langone Medical Ctr., New York, NY)

During conversation, speaker overlap is avoided while inter-turn silence is minimized. Notably, inter-speaker gaps are typically 200 ms or less—shorter than what is required to plan single words—suggesting that considerable speech planning (SP) occurs as speakers listen to their partners' turns. Though the psycholinguistic mechanisms of SP during turn-taking are well-studied, its neural dynamics are largely unknown. Using intracranial electrocorticography, we delineate SP-related activity from that of sensorimotor processes using questions (adapted from Bögels *et al.*, 2015) where the critical-information (CI) required to answer is provided at a singular timepoint. Consistent with involvement in SP, we observe that the left inferior frontal gyrus (IFG) is active immediately following CI. We also observe that IFG is preferentially active during SP but not general motor planning. Furthermore, we find that a subset of IFG planning sites are active during natural conversation while patients listen to the turns of opposing speakers. Finally, in preliminary experiments where IFG is stimulated while patients answer CI-questions, we find that IFG disruption results in significantly longer reaction times and lexical errors but not gross articulatory disruptions. In conclusion, we demonstrate that a subregion of IFG is critical for the SP processes employed during rapid conversational turn-taking.

3pSC8. Computational modelling of category stability in segment pairs participating in perceptual asymmetry. Ian Calloway (Linguist, Univ. of Michigan, 400 Lorch Hall, 611 Tappan Ave., Ann Arbor, MI 48109, iccalloway92@gmail.com)

Certain unidirectional sound changes show a similarity to the laboratory phenomenon of asymmetrical misperception. The unidirectionality of these processes mirror the dissimilar confusion rates of the two segments. Despite the similarity of these processes, it is not clear what role perceptual asymmetry plays in conditioning these changes. This study employs modeling to simulate change in the characteristics of consonant pairs whose confusion rates pattern with a unidirectional sound change: /k/-to-/t/ (before /i/) and /k/-to-/p/ (before /u/). Ten native AmEng speakers were recorded producing CVC words, where the initial consonant was /p/, /t/, or /k/ and the vowel varied in height and backness. Reduced and unreduced variants were elicited. Acoustically relevant features distinguishing /k/ from /t/ and /k/ from /p/ were identified using random forests. Reduced productions of /k/ and /p/ and /k/ and /t/, respectively, show higher acoustic similarity in vocalic contexts favoring increased confusability. This tendency toward acoustic similarity is predicted to condition category convergence. However, a language learner's category may be better informed by tokens that show less acoustic similarity to tokens of another category, predicted to condition divergence. Modelled results suggest that the stability of phonetic categories is sensitive to the relative weighting of these two forces.

3pSC9. HMM-based Bahasa Indonesia speech synthesis system with hand-segmentation and labeling. Elok Anggrayni (Dept. of Eng. Phys., Institut Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya, Indonesia 60111, Indonesia, anggrayni.elok@gmail.com), Dhany Arifianto (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia), and Joko Sarwono (Eng. Phys., Bandung Inst. of Technol., Bandung, Indonesia)

In this paper, we compare the naturalness quality of Bahasa Indonesia speech synthesis using festvox automatic- and hand-segmentation and labeling technique to create a speech transcription. First, we developed a 1549 declarative and question sentence phonetically balanced speech corpus uttered by six male and female speakers. We selected 47, 72, 119, 450, 929, and 1379 sentences, respectively for training whilst maintaining the phonetical balance. The objective is to find the least data training for synthesized naturalness evaluation on both automatic- and hand-segmentation and labeling. The evaluation result using the Mel-cepstrum distortion method was 2.9 for hand-segmentation and labeling, 5.36 for automatic with 47 training sentences, respectively which took about 45 minutes to complete. The

performance was increased by 2.46 with hand-segmentation and labeling, 4.78 for automatic, with 1379 sentences and about 9 hours of training time. The Mean Opinion Score was 3.98 (hand) and 3.04 for automatic, respectively which is about 18% performance improvement. The automatic segmentation and labeling introduced phoneme boundary errors which may suggest that the necessity to take careful consideration in segmentation and labeling.

3pSC10. Speaker tracking using SincNet for multi-channel naturalistic audio corpora: Apollo 11 fearless steps. Meena Chandra Shekar (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas at Dallas, 7220 McCallum Blvd, Apt. 1704, Dallas, TX 75252, Meena.ChandraShekar@utdallas.edu) and John H. L. Hansen (CRSS: Ctr. for Robust Speech Systems, Univ. of Texas at Dallas, Richardson, TX)

Apollo-11 was the first manned space mission to successfully bring astronauts to the moon and return them safely. As a massive collaborative effort, with astronauts flying the missions in outer space, the entire communications between flight controllers, their backroom support teams and, astronauts have taken place inside NASA Mission Control Center. The communications stemming from the experienced team of scientists, engineers, and technicians who worked collectively behind the scenes in the MOCR have not been identified. In this study, we propose to identify the speakers in the Mission Control Room for a small subset of 100 h derived from a collective 19 000 h of Apollo-11 audio data, which correspond to three challenging phases of the mission: lift-off, lunar landing, and lunar walking. The speaker identification solution explores a convolutional neural network architecture called SincNet across all three Apollo-11 phases. The speaker models obtained from the subset of 100 hours will be used to track speakers on the complete corpora. Our goal is to make it possible to be able to access each individual speaker involved with the Apollo 11 mission.

3pSC11. Accuracy of the language environment analysis (LENA) speech processing system for detecting communicative vocalizations of young children. Jaie C. Woodard, Nikaela Losievski, Meisam K. Arjmandi, Matthew Lehet (Michigan State Univ., East Lansing, MI), Yuanyuan Wang, Derek Houston (Ohio State Univ., Columbus, OH), and Laura Dilley (Dept. of Communicative Sci., Michigan State Univ., East Lansing, MI 48824, ldilley@msu.edu)

Automated audio processing systems, such as the Language Environment Analysis (LENA) system, are useful tools for understanding developmental language behaviors for clinical and basic research purposes. However, it is still unclear how accurate they may be in comparison to the traditional gold standard of evaluation by trained human listeners. In our study, human coders identified starts and ends of communicative vocalizations of children and adults from sampled audio in day-long LENA recordings of 23 families with a child with variable hearing status; accuracy of LENA was then determined for each recording by comparing LENA and human-derived labels for 100-ms frames of sampled audio. Preliminary analysis suggests that LENA accurately identified communicative vocalizations of the target child wearing the device as being produced by that target child 65% of the time (35% error); accuracy ranged from 49%—79% across recordings. When any child vocalization was correctly identified, LENA accurately distinguished whether this belonged to the target child or another child 75% of the time (25% error); accuracy, however, ranged from 7%—96%. These accuracy levels suggest caution is needed in applying popular speech processing systems like LENA to clinical and scientific questions in absence of additional validation measures.

3pSC12. Fearless steps: Taking the next step towards advanced speech technology for naturalistic audio. John H. L. Hansen (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080-3021, john.hansen@utdallas.edu), Aditya Joglekar, Abhijeet Sangwan, and Chengzhu Yu (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Dallas, Richardson, TX)

Over the past two decades, machine learning technologies have been targeting real-world problems in the Speech-Language (SLT) domain. Speech corpora developed under diverse environments have been paramount to progress, though most are simulated/controlled scenarios. Success in Machine Learning for SLT requires new innovative challenges posed by multi-speaker naturalistic audio. The UTDallas-CRSS led Fearless Steps (FS) initiative over the past 6 years has made significant strides through the development of the Fearless Steps Corpus, and multiple novel schemes to address core speech tasks. The next steps taken through this initiative aim at motivating further research on these data through worldwide collaborative efforts. To achieve this, the Inaugural Fearless Steps Challenge held in 2019 saw the release of 11 000 h of Apollo-11 audio data and diarization transcripts, made freely available to public online. An additional 100 h of manually annotated data was released as a Challenge corpus, available to researchers interested in working on any of the five core speech tasks: Speech-Activity-Detection, Speaker Diarization and Identification, Speech Recognition, and Sentiment Detection. FS Challenge resulted in over 150 participants worldwide developing novel task-specific algorithms. Going forward, the FS initiative aims at digitizing and releasing over 150 000 h of the remaining Apollo Missions in conjunction with follow-on Challenge Tasks focused on developing multi-channel and conversational speech systems.

3pSC13. Fearless steps, NASA's first heroes: Conversational speech analysis of the Apollo-11 mission control personnel. Aditya Joglekar (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080-3021, aditya.joglekar@utdallas.edu) and John H. L. Hansen (Dept. of Elec. and Comput. Eng., The Univ. of Texas at Dallas, Richardson, TX)

Between 1963 and 1972, a massive team of dedicated scientists, engineers, and specialists at the NASA Mission Control Center (MCC) worked seamlessly together in a cohesive manner to successfully carry out multiple manned missions to the moon. All communications between personnel were carried out over multiple inter-connected audio channels and recorded on two 30-track analog tapes. Digitization of the entire Apollo-11 mission tapes made possible through the UTDallas-CRSS Fearless Steps (FS) initiative contains the recordings of all the crew members including the three astronauts. With over 600 speakers in constant communication ensuring the astronauts' safety throughout the mission, the success of the Apollo missions can be attributed largely to the MCC crew members. The focus of this effort is thus, to analyze their high-level group dynamics and intricate communication characteristics and gain insight into the success parameters involved in such large-scale time-critical operations. The 100 hour FS Challenge corpus highlights multiple salient moments like Lift-Off and Lunar-Landing and poses its' own challenges for core speech tasks. This effort aims at analyzing spontaneous multi-speaker conversations in recordings often degraded by various noise types. Analysis of speech characteristics under varying stress, overlap, and noise conditions are observed to develop novel speech-activity and speaker models. These domain-specific models are leveraged to develop state-of-the-art SLT systems.

3pSC14. Deep learning model for automated assessment of lexical stress of non-native english speakers. Daniel Korzekwa (Amazon, Jana Pawla II 3a/9, Gdansk 80-462, Poland, korzekwa@amazon.com) and Bozena Kostek (Audio Acoust. Lab, Gdansk Univ. of Technol., Gdansk, Poland)

In this paper, we present a novel system to practice lexical stress in L2 English learning with Amazon Alexa home assistant. The language learning for non-native English speakers mostly focuses on practicing correct grammar, extending language vocabulary, and improving pronunciation. The system proposed enables a person to practice lexical stress skills at home by having conversations with Alexa assistant. The system assesses student's abilities to enunciate words with a correct lexical stress and automatically selects the next words to practice. After a series of exercises, the system informs the student on the improvement. The main scientific contribution of the work presented is a deep learning model for automated assessment of lexical stress of non-native English speakers. The model is based on a transfer learning technique. First, we train the model to predict the location of a lexical stress on a syllable level using a large corpus of native English speech. Then, we tune the model with a limited amount of a non-native speech. A corpus of non-native English speech obtained from Polish speakers is incorporated into the training and testing of the model. It is

shown that the system enables to create a vocabulary for a particular speaker interactively.

3pSC15. Automatic perceptual judgment using neural networks. Seong-jin Park (Dept. of Linguist, Univ. of Arizona, Tucson, AZ 85721, seongjinpark@email.arizona.edu) and John Culnan (Univ. of Arizona, Tucson, AZ)

The aim of the present study is to investigate the performance of automatic perceptual judgment models built with neural networks. In previous studies, Franco *et al*. (1997) used HMM-derived scores based on posterior probabilities of phone segments, demonstrating a high correlation with human raters. Deville *et al*. (1999) also used an HMM/ANN recognition approach, and showed how the results of automatic speech recognition can be used for perceptual judgments. However, since most previous studies made use of automatic speech recognition in their analysis, the present study provides a different approach: using features and raw data. Native speakers of English will listen to English sentences produced by native and non-native speakers of English, transcribe what they heard, and respond to one of three perceptual judgements: foreign-accentedness, fluency, and comprehensibility. The data will be fed into prediction models in three different ways; one with annotated features (pauses, durations, etc), another with Mel Frequency Cepstral Coefficients (MFCC), and the other with Mel-spectrograms. The performance of the models will be measured by analyzing the correlation between the judgments by models and by human raters. The preliminary results of this study will be used to build more accurate automatic proficiency judgment models.

3pSC16. Improved vowel labeling for prenasal merger using customized forced alignment. Yuanming Shi (Inst. for Artificial Intelligence, Univ. of Georgia, Athens, GA 30602, jeremy.shi@uga.edu), Margaret E. Renwick (Dept. of Linguist, Univ. of Georgia, Athens, GA), and Frederick Maier (Inst. for Artificial Intelligence, Univ. of Georgia, Athens, GA)

Forced alignment is a popular technique for gaining phone-level audio transcriptions, but the pronunciation dictionaries used by it are typically based on standard varieties of US English, leading to errorful outputs for non-standard varieties. We employ a customized pronunciation dictionary with the Montreal Forced Aligner to increase labeling accuracy of the prenasal merger (a.k.a. pin-pen merger) in Southern US English. We allow the aligner to choose between IH (/ɪ/) and EH (/ɛ/) in words where the merger is expected, rather than enforcing a standard, unmerged pronunciation. We examine the tokens reclassified from EH to IH when using the new dictionary, and we use formant values to study the acoustic separation (measured by Pillai scores and Euclidean distances between centroids) between vowel formant clusters. When applied to the Digital Archive of Southern Speech (DASS), we find that the modification increases the separation between the prenasal allophones of IH and EH, and also that the proportion of prenasal EH tokens reclassified to IH is correlated with the original degree of separation between prenasal IH and EH for each DASS speaker. K-means clustering is also used to show the modification yields more accurate phonetic transcriptions, measured by increased precision and recall.

3pSC17. Accurate reading rate: Validations of machine scoring. Masanori Suzuki (Analytic Measures, Inc., Palo Alto, CA), Jared Bernstein (Analytic Measures, Inc., 1330 Tasso St., Palo Alto, CA 94301, jared413@stanford.edu), Jian Cheng, and Tomoyo Okuda (Analytic Measures, Inc., Palo Alto, CA)

Accurate oral reading rate is a common measure of early reading proficiency (grades K-4). A concurrent validation study compared a fully automated measure of accurate reading rate (words read correctly per minute) to DIBELS, a well-established, human-scored assessment. We report data from 174 students at a parochial school in Delaware and 130 students at a public school in Texas. Each student took three forms of an automatically scored test (yielding 898 test forms automatically scored) and also took the grade-appropriate form of the standard DIBELS test, which was scored by a trained person. Test administration order was counterbalanced. Correlations were computed for Machine-Machine and Human-Machine score pairs, for each grade separately and for the total dataset. When all grades are combined, Pearson $r = 0.87$, and correlations for single grades are all above 0.85, thus at or above the reliability ceiling of the DIBELS criterion score. Also, the automated tests have higher test-retest reliability ($r = 0.91$) than reported in an independent DIBELS study (Goffreda and DiPerna, 2010). [This research was supported by IES, U.S. Department of Education, via contract ED-IES-17-C-0030 to Analytic Measures, Inc. Disclosure: This report of an accurate automatic method may potentially benefit the authors financially in the future.]

Session 3pSP

Signal Processing in Acoustics: General Topics in Signal Processing (Poster Session)

Kay L. Gemba, Chair
MPL/SIO, UCSD, University of California, San Diego, 8820 Shellback Way, Spiess Hall, Room 446, La Jolla, California 92093

All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Contributed Papers

3pSP1. Robust neural network approach to source-range estimation in a simulated Arctic environment. Rui Chen (Mech. Eng., MIT, 77 Massachusetts Ave. Rm. 5-223, Cambridge, MA 02139, ruic@mit.edu) and Henrik Schmidt (Mech. Eng., MIT, Cambridge, MA)

Underwater source-range estimation is conventionally performed via matched field processing (MFP), which relies on precise modelling of the propagation environment to produce accurate estimates. Here, we design a convolutional neural network (CNN) for long range (up to 50 km) source-range estimation in a complex Arctic propagation environment and compare its performance and robustness to MFP. The model parameters are optimized through cross-validation with regularization and other precautions implemented to deter overfitting. It is trained and tested with simulated data recorded on a vertical line array and its robustness to environmental mismatch is examined by introducing deviations to the original sound speed profile (SSP) when generating the test data. Results show that the CNN is more robust to environmental changes than MFP but at the expense of worse performance when the environmental parameters are accurately modelled. Insights into how the CNN performs range estimation is discussed as well. [Work supported by Office of Naval Research.]

3pSP2. Application of autoencoder to traffic noise analysis. Andrzej Czyzewski (Gdansk Univ. of Technol., Narutowicza 11/12, Gdansk 80-233, Poland, ac@pg.gda.pl), Adam Kurowski, and Szymon Zaporowski (Gdansk Univ. of Technol., Gdansk, Poland)

The aim of an autoencoder neural network is to transform the input data into a lower-dimensional code and then to reconstruct the output from this code representation. Applications of autoencoders to classifying sound events in the road traffic have not been found in the literature. The presented research aims to determine whether such an unsupervised learning method may be used for deploying classification algorithms applied to the automatic annotation of road traffic-related events based on noise analysis. Two-dimensional representation of traffic sounds based on Mel Frequency Cepstral Coefficients (MFCC) was fed the autoencoder neural network, and after that classified with k-nearest neighbors algorithm, Support Vector Machines, and random forests. Obtained results show that sound recordings can help determine the number of vehicles passing on the road. However, instead of being treated as independent, this method output should be combined with another source of data, e.g., video processing results or microwave radar data readings. Comparative results of vehicle counting obtained with the use of autoencoder and different classifiers are shown in the paper. [The Polish National Centre finances the project for Research and Development (NCBR) from the European Regional Development Fund No. POIR.04.01.04-00-0089/16 entitled: "INZNAK: Intelligent Road Signs with V2X Interface for Adaptive Traffic Controlling."]

3pSP3. Production and perception of English vowels preceding voiced and voiceless consonants by Korean learners of English. Juyeon Chung (Linguist, Indiana Univ., Memorial Hall 322, Bloomington, IN 47405-7005, chungjulia29@gmail.com)

In English, consonant voicing has large effects on the previous vowel duration, as does the status of the vowel as tense or lax. Our aim is to examine whether there is an interplay between L1 interference and L2 experience on L2 English vowel productions and perceptions. Korean L2 speakers were chosen as participants; Korean has no contrasts in tense versus lax vowels, and no coda consonant voicing contrast in a monosyllabic structure, but exhibits post-vocalic voicing contrasts in disyllabic structures, and in these cases, voiced consonants exhibit the lengthening of the preceding vowel. The participants were asked to read the list of English nonce words consisting of the English high and mid vowels and sets of plosives contrasting in voicing as a coda in the two different structures. In an ABX discrimination task, the subjects were asked to identify the different words from among minimal triples. None of the speakers exhibited different patterns in both structures in productions or perceptions. All speakers exhibited durational correlates to the voicing contrast, and to the tense-lax distinction. The effect of L2 English experience was not shown in perception task. Overall, there was no correlation between vowel length differences in production and perception scores.

3pSP4. Interpreting the latent representations of a convolutional neural network trained on spectrograms. Mark Thomas (Faculty of Comput. Sci., Dalhousie Univ., 6050 University Ave., PO BOX 15000, Halifax, NS B3H 4R2, Canada, mark.thomas@dal.ca), Bruce Martin, Katie Kowarski, Briand Gaudet (JASCO Appl. Sci., Dartmouth, NS, Canada), and Stan Matwin (Faculty of Comput. Sci., Dalhousie Univ., Halifax, NS, Canada)

Recent work [1,2] has shown that Convolutional Neural Networks (CNNs) trained on spectrograms of acoustic signals are capable of learning high-level latent representations for the purpose of detecting and classifying the vocalizations of endangered baleen whales. The aforementioned latent representations were used in the development of an automated system that was capable of detecting the vocalizations of blue, fin, and sei whales against non-biological and ambient noise sources to a high degree of accuracy (0.961, F-1 Score = 0.899). In this work, we conduct an exploratory analysis of the same latent representations using statistical machine learning approaches as well as by visualizing the convolutional feature maps learned by the CNN. Through this analysis we attempt to interpret what properties of a spectrogram are easily and/or most often exploited by the CNN during training in order to improve upon the state-of-the-art and develop more robust detection systems going forward. [1] M. Thomas, B. Martin, K. Kowarski, B. Gaudet, and S. Matwin, *Marine Mammal Species Classification using Convolutional Neural Networks and a Novel Acoustic Representation, ECML PKDD 2019* (Springer, Cham, 2019). [2] M. Thomas, "Towards a

novel data representation for classifying acoustic signals, in *Canadian Conference on Artificial Intelligence* (Springer, Cham, 2019).

3pSP5. An end-to-end approach for true detection of low frequency marine mammal vocalizations. Mark Thomas (Faculty of Comput. Sci., Dalhousie Univ., 6050 University Ave., PO BOX 15000, Halifax, NS B3H 4R2, Canada, mark.thomas@dal.ca), Bruce Martin, Katie Kowarski, Briand Gaudet (JASCO Appl. Sci., Dartmouth, NS, Canada), and Stan Matwin (Faculty of Comput. Sci., Dalhousie Univ., Halifax, NS, Canada)

Research into automated systems for detecting marine mammal vocalizations within acoustic recordings is expanding internationally due to the necessity to analyze large collections of data collected for passive acoustic monitoring. Recent work towards the development of such systems using Convolutional Neural Networks (CNNs) shows great promise and these systems are capable of generalizing to additional species without having to re-train the entire network [1]. However, to the best of our knowledge, the current deep learning implementations do not perform what we refer to as *true* detection. Instead these systems are simply capable of determining the presence or absence of a vocalization within a spectrogram. In this work we present a CNN trained on spectrograms containing labelled bounding boxes around low-frequency vocalizations produced by several species of marine mammals. In this way, the CNN can precisely detect vocalizations in terms of both time and frequency, while maintaining the advantage of being generalizable to additional species. [1] M. Thomas, B. Martin, K. Kowarski, B. Gaudet, and S. Stan, Marine mammal species classification using convolutional neural networks and a novel acoustic representation, in *ECML PKDD 2019* (Springer, Cham, 2019).

3pSP6. Frequency-based multiband adaptive compression for hearing aid application. Kashyap Patel (Dept. of Elec. and Comput. Eng., Univ. of Texas at Dallas, ECSN 4218, 800 W Campbell Rd., Richardson, TX 75080, kxp180006@utdallas.edu), Nikhil Shankar, and Issa M. Panahi (Dept. of Elec. and Comput. Eng., Univ. of Texas at Dallas, Richardson, TX)

Multiband Dynamic Range (MBDR) Compressor is the heart of signal processing in hearing aid devices (HADs). Operating speed of the MBDR compressor plays an important role in preserving the quality and intelligibility of the output signal. Traditional fast-acting compressor preserves the audible cues in quiet speech but, in presence of surrounding noise, it can degrade the sound quality by introducing pumping and breathing effects. Alternatively, slow-acting compressor maintains the temporal cues and the listening comfort but may provide inadequate gain for soft inputs that come right after loud inputs. HADs may operate in a variable acoustic environment. Therefore, a fixed speed in compression might affect the performance of the hearing aids. In this study, we propose a frequency (FFT) based nine-band adaptive MBDR compression which uses spectral flux as a measure of the intensity change in input level to adapt the speed of the compressor in each band. Gain, threshold and compression ratio of the compressor for nine bands are adjusted based on the audiogram of the hearing impaired patient. The proposed frequency-based adaptive MBDR compression method is implemented on smartphone. The objective and subjective test results demonstrate the performance of proposed method compared to fixed compression approach in different acoustic environments.

3pSP7. Investigation of obstacle avoidance algorithm in paired-driving autonomous mobile robots revealed by mimicking ultrasonic sensing in bats. Tomoya Kubota (Graduate School of Life and Medical Sci., Doshisha Univ., 1-3, Tatara Miyakodani, Kyotanabe 610-0321, Japan, ctud1013@mail4.doshisha.ac.jp), Shoya Nakade (Graduate School of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan), Yasufumi Yamada (Programs of Mathematical and Life Sci., Hiroshima Univ., Hiroshima, Japan), Takaaki Asada, Shinichi Sasaki (Murata Manufacturing, Kyoto, Japan), and Shizuko Hiryu (Graduate School of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan)

To evaluate the biosonar behavior of bats in paired flight from an engineering point of view, obstacle avoidance driving experiments were conducted using two autonomous mobile robots embedded with one ultrasound transmitter and two receivers. It has been reported that when two bats aim at the same prey simultaneously, the following bat temporarily stops the pulse emission and tracks the preceding bat. It is supposed that this is to avoid interference with signals by other individuals, i.e., utilizing pulses emitted by other individuals and the echoes for their own sensing. Then we proposed the hybrid sensing which switches active and passive sensing according to the situation, and the completion rate of obstacle route in the paired-driving robots was examined. As the result, it was proven that hybrid sensing robot reduced the emission frequency of the pulse, but the completion rate of the obstacle route was high in comparison with case of the paired-driving robots with active sensing. In addition, in order to improve localization performance and to emit FM type ultrasonic pulse of bats, thermoacoustic transducer (thermophone) with broadband frequency characteristics are introduced as a ultrasound transmitter. [Work supported by JSPS-KAKENHI;JP18H03786, JP16H06542 and JST-PESTO.]

3pSP8. Alignment of canonical and realized acoustic cue labels for modification patterns in speech analysis. Christine Soh (Speech Commun. Group, RLE, MIT, 405 Memorial Dr., Cambridge, MA 02139, csoh@mit.edu), Tanya Talkar, Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (Speech Commun. Group, RLE, MIT, Cambridge, MA)

Acoustic cues are properties of the speech signal that provide information about the distinctive features of the speaker's intended words and phonemes. Analysis of acoustic cues can indicate reductions and modifications in speech, in which landmarks and other feature cues are deleted, inserted, or substituted for others, and can be informative in distinguishing underlying causes of speech impairments. To extract this information about modifications, we need to determine which predicted canonical labels the realized labels correspond to. We propose an algorithm that uses a time-based alignment method for the landmarks as well as a modified labeling scheme to more accurately find correspondences between realized landmarks and distinctive features to the canonical labels. The results show improved alignment not only for the realized landmark labels but also for the labels of other feature cues, enabling accurate and holistic analysis of modifications in speech, at the more detailed level of cues to distinctive features rather than the phoneme or phone level. Using this algorithm, we analyze a database of CN-REP (Children's Nonword Repetition Task) recordings from children diagnosed with speech impairments and find several potential modification markers that may distinguish among different diagnoses.

3pSP9. A computationally efficient blind source separation for hearing aid applications and its real-time implementation on smartphone. Gautam Shreedhar Bhat (Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080, gxs160730@utdallas.edu), Nikhil Shankar, and Issa Panahi (Univ. of Texas at Dallas, Richardson, TX)

Conventional Blind Source Separation (BSS) techniques are computationally complex. This is due to the calculation of the demixing matrix for the entire signal or due to the frequent update of the demixing matrix at every time frame index, making them impractical to use in many real-time applications. In this paper, a robust, neural network based two-microphone sound source localization method is used as a criterion to enhance the efficiency of the Independent Vector Analysis (IVA), a BSS method. IVA is used to separate speech and noise sources which are convolutedly mixed. The practical usability of the proposed method is proved by implementing it on a smartphone to separate speech and noise in real-world scenarios for hearing-aid applications. The experimental results with objective and subjective tests reveal the usefulness of the developed method for real-world applications.

3pSP10. Fast parameter estimation of distributed sources in shallow water with a new UCA. Qingyue Gu, Huigang Wang (School of Navigation, Northwestern PolyTech. Univ., Xi'an, Shaanxi, China), Weitao Sun (School of Navigation, Northwestern PolyTech. Univ., 127 West Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, sunwt1223@Gmail.com), and Yifeng Xu (School of Navigation, Northwestern PolyTech. Univ., Xi'an, Shaanxi, China)

Due to large number of reflections from surface and bottom in the shallow ocean environment, the radiated noise from the same source can be the distributed source model, instead of the traditional point model. The distributed source will appear two-dimension spatial distribution characteristic in azimuth and elevation. Four important parameters of two-dimensional spatial distribution source are the central angle and extended angle in both dimension, which is very important to identify the type of source. In this paper, a novel UCA array layout is proposed for an two-dimensional incoherent distributed(TDID) source, and a new ESPRIT algorithm for two-dimension distribution source is derived to estimate these parameters analytically with symmetrical structure of UCA. A large mount simulation of underwater environment shows this method can achieve high computational accuracy with low computational burden, and the results also verify the robustness to the different distribution forms and good performance with larger extended angle size.

3pSP11. Target detection in pseudo Wigner-Ville distribution of underwater beamformed signals. Yeon-Seong Choo (Dept. of Ship and Ocean Eng., Korea Univ. of Sci. and Technol., 32 Yuseong-daero 1312beon-gill, Yuseong-gu, Daejeon 34103, South Korea, chooys@kriso.re.kr), Sung-Hoon Byun, Sea-Moon Kim (Marine ICT Res. Div., Korea Res. Inst. of Ships and Ocean Eng., Daejeon, South Korea), and Keunhwa Lee (Sejong Univ., Seoul, South Korea)

Because of clutter or reverberation in the water, it is difficult to distinguish a target from them with a beamforming image. It is well-known that the pseudo Wigner-Ville distribution, which is high-resolution time-frequency processing, can detect target signals effectively. In this study, the pseudo Wigner-Ville distribution with the beamformed signals was analyzed to detect an underwater target. The acoustic signals were measured using an array system consisting of a transmitting transducer and a line array of 16 elements at sea. The target was an air-filled aluminum spherical shell with a thin thickness. The transmitted signal was 3 to 8 kHz linear frequency modulation continuous waveforms. The experiment results show that the target can be easily identified with the pseudo Wigner-Ville distribution. [Work financially supported by the research project PES3180 funded by KRISO.]

3pSP12. Speaker identification using convolutional-long short-term memory neural networks. Serkan Tokgoz (Dept. of Elec. Eng., The Univ. of Texas at Dallas, EC33, 800 West Campbell Rd., Richardson, TX 75080, sxt167830@utdallas.edu) and Issa M. Panahi (Elec. Eng., The Univ. of Texas at Dallas, Richardson, TX)

Speaker identification (SI) techniques has been used in numerous commercial products over the last decades. In SI, the main purpose is to match a voice sample from an unknown speaker to one of the labeled speaker models. To be able accomplish this task, there are two operational phases, training (can be also termed as enrollment) and testing. In both phases, feature extraction and feature matching are the two key steps. In this work, we have extracted features with Mel-Frequency Cepstrum Coefficients (MFCC) because MFCC has accurate representation of the vocal tract, and a Spectral-Flux based Voice Activity Detector (VAD) is implemented to extract features from the speech segments. In feature matching task, we build a Convolutional Long short-term memory (LSTM) Neural Network for the speaker models. We examine main performance of the system in terms of identification rate and compare the proposed method with other SI methods under several noisy conditions at different signal to noise ratio (SNR) levels.

3pSP13. Deep neural network based direction of arrival estimation for hearing aid applications using smartphone. Abdullah Küçük (Elec. and Comput. Eng., The Univ. of Texas at Dallas, 800 West Campbell Rd., Richardson, TX 75080, axk166230@utdallas.edu) and Issa M. Panahi (Elec. and Comput. Eng., The Univ. of Texas at Dallas, Richardson, TX)

Deep neural network (DNN) techniques are gaining popularity due to performance boost in many applications. In this work we propose a DNN-based method for finding the direction of arrival (DOA) of speech source for hearing aid applications using smartphones. We consider the DOA estimation as a classification problem and use the magnitude and phase of speech signal as a feature set for DNN training stage and obtaining appropriate model. The model is trained and derived using real noisy speech data recorded on smartphone in different environments under low SNRs. The DNN-based DOA method, with the pre-trained model, is implemented and run on Android smartphone in real time and evaluated objectively and subjectively. The test results are presented showing the performance of proposed method versus other methods.

3pSP14. Modeling acoustic cues to distinctive features in a lexical speech analysis system. Hoang Nguyen (Elec. Eng. and Comput. Sci., MIT, 974 White Knoll Dr. Apt. 17, Los Angeles, CA 90012, hoangn@mit.edu), Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (Res. Lab. of Electronics, MIT, Cambridge, MA)

Acoustic cues are robust elements that can be used to infer information contained in the speech signal, such as underlying linguistic distinctive features and the words intended by the speaker (Stevens *JASA* 2002). Yet, most current automatic speech recognition systems do not take advantage of a feature-cue-based framework for signal analysis. In this project, a set of common acoustic cues has been explicitly modeled by Gaussian mixture models. This set of acoustic cues can provide evidence for the overall phoneme and word sequences of an utterance. The extracted cues and their values can also determine a speaker's linguistic production pattern, i.e., the systematic context-governed modifications in surface-phonetic form that occur pervasively in conversational speech. The simple Gaussian mixture model representation structure reduces the need for extensive amounts of training data, in contrast to conventional schemes based on large neural networks.

3pSP15. Self-fit generation of the wide range compression parameters in hearing aids. Apurba Bose (Elec. and Comput. Eng., Univ. of California San Diego, 815 Rita Atkinson Gilman Dr., La Jolla, CA 92092, apbose@eng.ucsd.edu), Ziqi Gan, and Harinath Garudadri (Univ. of California San Diego, La Jolla, CA)

OpenMHA is an open source software executing the real time signal processing algorithms with reduced latency between the input and the output signals. In this abstract, we elaborate on a hearing aid fitting algorithm, which tunes the Wide Dynamic Range Compression (WDRC) algorithm parameters through a A-B comparison test. The following approach is adopted-The left and the right audiogram values are clustered using the Gaussian Mixture model. A value is assigned to more than one cluster with a probability. The cluster labels are assigned according to the model yielding the maximum log likelihood-The best fit for the audiogram values of user is used to calculate the WDRC parameters. Recording the audiogram values of individual user and tuning the parameters by the audiologists can be a cumbersome task. Web-app is implemented to present A-B comparison to the users to determine the best fit audiogram values. The Open Speech Platform (OSP) comprises of Embedded Web Server (EWS) presenting the web-app. It guides the user to determine the best fit through a AVL tree based binary search algorithm. The device aims to determine the best fit in reduced number of steps, post which the WDRC parameters are determined through the NAL-NL2 libraries. This search algorithm for each user, and listener's feedback on the success of the fit using the ecological monetary assessment (EMA) app, with the environmental sound data logging can be leveraged further to create auto fit algorithms in dynamic environments using MCTS in reinforcement learning.

Session 3pUWa

Underwater Acoustics: Source and System Component Localization

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Contributed Papers

1:00

3pUWa1. A deep learning approach to source localization and seabed classification using pressure time-series from a vertical array. Mason C. Acree (Phys., Utah Valley Univ., 4142 N Heatherfield Ln, Lehi, UT 84043, mason7acree@gmail.com), David F. Van Komen, Tracianne B. Neilsen (Phys. & Astronomy, Brigham Young Univ., Provo, UT), and David P. Knobles (KSA, LLC, Austin, TX)

Source localization in a shallow ocean environment has historically been done using optimization techniques such as matched-field processing. However, such optimizations depend on the parameterization of the ocean environment. Due to the complexity of this physical system, some researchers are currently applying machine and deep learning techniques to source localization problems. We propose a convolution neural network (CNN) to better predict the source localization and seabed classification simultaneously using pressure time series waveforms from a vertical line array. Building on research using a CNN to classify the source locale and seabed type using waveforms from only one hydrophone, the method has been extended to a 16-element vertical line array. The additional hydrophones add more physical information from the system as well as simply more features for a CNN to learn source range, depth, and seabed type. The synthetic data were generated using a range independent normal-mode model for multiple ocean environments. Modifications to the CNN are made to exploit the multi-channel waveforms. Ocean acoustic applications require this accurate classification of source locale and seabed environment. In future work we will extend our technique and CNN model to work with real world data.

1:15

3pUWa2. Underwater object localization using compressive sensing. Margiasih P. Liana (Eng. Phys., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia), Dhany Arifianto (Dept. of Eng. Phys., Institut Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id), and Wirawan Wirawan (Elec. Eng., Institut Teknologi Sepuluh Nopember, Surabaya, Indonesia)

In this paper, we propose a technique to localize a moving submerged object with a sparse measurement using a hydrophone array. We used a natural muddy pond bottom with 9 m length, 5.5 m width, and 2 m depth. First, we measured the pond impulse response to identify the sound propagation characteristics. In the second phase namely static direction of arrival (DOA), the underwater speaker was placed at half-circle positions with respect to the array to simulate the underwater object trajectory, 0 deg, 30 deg, 60 deg, 90 deg, 120 deg, 150 deg, and 180 deg, respectively. The sound generated by a four-blade propeller submarine toy was measured to track its trajectory as a dynamic DOA measurement. The measured data were transformed into discrete cosine transform coefficients and reconstructed sparsely by using the basis pursuit algorithm. Based on the estimated incident angle and time delay, the reconstructed measurements were then compared to full dictionary measurement. The results showed that the error angle of the origin deviated about 2.5 deg with half of the measured data were lost. This may suggest that the sound propagation was degraded due to bottom reverberation and scatterers. Currently, we do a similar measurement in an open shallow water environment to eliminate the reverberation.

1:30

3pUWa3. Using relative channel impulse responses to train a machine learning algorithm for 2-D surface ship source localization. Nicholas C. Durofchalk (Mech. Eng., Georgia Inst. of Technol., 2788 Defoors Ferry Rd., Apt 325, Atlanta 30318, Georgia, ndurofchalk3@gatech.edu), Arslan Ali, Saibal Mukhopadhyay, Justin Romberg (Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA), and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Underwater source localization using receiver arrays is often achieved with a purely model-based approached, i.e., matched-field processing using simulated replica-field. However, these approaches only yield reasonable predictions if the complicated ocean environment can be precisely described, which is often a daunting task. Alternatively, by turning to a data-driven approach, source localization can be achieved without the strong dependence on model-parameters. Recently, source localization in the Santa Barbara Channel (~580 m depth, downward refracting profile) has been achieved via measurements of ships of opportunity, and machine learning classifiers. In this presentation, relative channel impulse responses, estimated via the ray-based blind deconvolution, are used to train a neural network to predict the latitude and longitude of a surface ship within the Santa Barbara Channel. Localization results for simulated and experiment data are presented and implications for future work are discussed.

1:45

3pUWa4. Effect of signal to noise ratio on a convolutional neural network for source ranging and environmental classification. Kira Howarth (Dept. of Phys., Brigham Young Univ., BYU N283 ESC, Provo, UT 84602, howarthke@gmail.com), David F. Van Komen, Tracianne B. Neilsen (Brigham Young Univ., Provo, UT), David P. Knobles (KSA, LLC, Austin, TX), Peter H. Dahl, and David R. Dall'Osto (Univ. of Washington, Seattle, WA)

In ocean acoustics, simultaneous estimation of both source-receiver range and environment are complicated by low signal-to-noise ratio (SNR). Range and environment class can be found with a convolutional neural network (CNN), which is chosen because of its ability to find patterns in grid-structured data. The CNN acts on synthetic pressure time series data from a single receiver generated for four canonical environments: deep mud, mud over sand, sandy silt, and sand. Data were split into training and validation sets. The CNN is trained to identify source range and environmental class. The change in performance for different SNR values is evaluated by adding Gaussian-distributed noise. A study is done regarding the impact of having different SNR values for the training and validation datasets. The trained

CNN is applied to pressure time series data measured on the APL-UW Intensity Vector Autonomous Receiver system at SBCEX. This study shows that performance depends more on the suitability of the training dataset than on the SNR value, implying that a CNN has potential to both estimate range and environmental class, even when there is low SNR. [Work supported from Office of Naval Research.]

2:00

3pUWa5. Ocean source localization with multi-frequency deep learning in uncertain environments. Haiqiang Niu (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Beijing 100190, China, nhq@mail.ioa.ac.cn), ZaiXiao Gong (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China), Emma Reeves Ozanich, Peter Gerstoft (Scripps Inst. of Oceanogr., La Jolla, CA), Haibin Wang, and Zhenglin Li (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

A deep learning approach is proposed to locate broadband acoustic sources in ocean waveguides with uncertain bottom parameters. The residual neural networks, trained on a huge number of sound field replicas generated by an acoustic propagation model, are used to handle the environmental uncertainty in source localization. A two-step training strategy is presented to improve the training of the deep models. First, the range is discretized in a coarse (5 km) grid. Subsequently, the source range within the selected interval and source depth are discretized on a finer (0.1 km and 2 m) grid. The deep learning methods are demonstrated for simulated magnitude-only multi-frequency data in uncertain environments. Experimental results from the China Yellow Sea show that the approach is comparable with SAGA in performance while much faster in computation speed.

2:15

3pUWa6. Robust matched field processing for array tilt and environmental mismatch. Gihoon Byun (Scripps Inst. of Oceanogr., La Jolla, San Diego, CA 92093-0238, gbyun@ucsd.edu), Hunter Akins, Hee-Chun Song, and William A. Kuperman (Scripps Inst. of Oceanogr., La Jolla, CA)

Adaptive matched field processing with the minimum variance beamformer provides excellent sidelobe suppression for source localization, but suffers from sensitivity to mismatch between the modeled and true acoustic field (i.e., environmental mismatch). To increase tolerance to the mismatch while retaining satisfactory sidelobe control, robust algorithms such as the white noise constraint (WNC) can be employed. The WNC alone, however, is not sufficient when the mismatch results from an unknown array tilt (i.e., geometric mismatch). This study introduces an adaptive matched field beamformer that is tolerant to both array tilt and environmental mismatch. By modeling the pressure fields corresponding to a set of assumed tilt angles, we impose multiple constraints that, when applied to the beamformer, increase robustness to the array tilt. Simulations and data results are presented to demonstrate localization and tracking of a surface ship (200–500 Hz) using a 16-element, 56-m long, tilted vertical array in approximately 100-m deep shallow water.

2:30

3pUWa7. Investigation on one-step integrated positioning method for seafloor control network localization. Shuang Zhao (China Univ. of Petroleum (East China), No. 66, Changjiang West Rd., Huangdao Dist., Qingdao, Shandong 266580, China, zsunique.vip@gmail.com) and Zhenjie Wang (China Univ. of Petroleum (East China), Qingdao, China)

Global Navigation Satellite System-Acoustic (GNSS-A) combined positioning is the main technique for seafloor geodetic control points localization, typically including datum transfer and subsequent positioning adjustment. This conventional step-wise method is time-consuming and accuracy-limited. In this paper, a novel one-step integrated positioning method for seafloor control network localization is proposed, which estimates positions of surveying vessels and seafloor points simultaneously. The effectiveness of new method is verified by computer simulation and comparative analysis. First, the conventional underwater acoustic positioning stepwise adjustment method is introduced and the positioning errors are analyzed. Then, to better constrain positions, the baseline lengths and depth-differences between pair of seafloor transponders are taken into consideration. The novel integrated positioning model based on overall adjustment is established. Furthermore, to deal with rank deficiency problem in overall adjustment, selecting weight fitting (SWF) method is applied in consideration of reliable priori information of surveying vessel coordinates. Finally, the regulating parameter determination for SWF method is discussed. Tests demonstrate that depth-differences constraint performs better than baseline length constraint in light of vertical geometry weakness. The novel method is notably superior to the conventional method in terms of positioning accuracy according to rms (Root Mean Square) statistics.

Session 3pUWb

Underwater Acoustics: Underwater Acoustics Topics (Poster Session)

Timothy F. Duda, Cochair
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All posters will be on display from 1:00 p.m. to 4:00 p.m. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 2:30 p.m. and authors of even-numbered papers will be at their posters from 2:30 p.m. to 4:00 p.m.

Contributed Papers

3pUWb1. Acoustic camera modeling and compensation technique based on SONAR equation. Ho Seuk Bae (Agency for Defense Development, Jinhae P.O.Box 18, Jinhae-gu, Changwon-si 51678, South Korea, belfre@add.re.kr) and Woo-Shik Kim (Agency for Defense Development, Changwon-si, South Korea)

Acoustic camera is a type of ultrasonic equipment used to detect or classify nearby objects located in water. The distortion of SONAR images acquired from conventional acoustic camera equipment inevitably occurs depending on the platform speed and operating frequency. This has not been huge concern because the platform speeds were low. However, more recently, a high accuracy of acoustic camera images has become essential in accordance with the development of unmanned autonomous systems. Additionally, a high speed of operating platform is also required. Therefore, there is a need for the study of image correction to reduce the distortion of SONAR images acquired from the equipment. In this study, we simulate virtual acoustic camera SONAR images by performing numerical modeling based on the SONAR equation to improve the accuracy of acoustic camera images. To verify the validity of the simulated images, we use the literature data and water-tank experiment data. Finally, we conduct the image compensation both in the direction of motion relative to the platform speed, and in the transverse direction relative to the platform speed and operating frequency.

3pUWb2. Automatic detection and classification of sounds generated by ocean vehicles via cyclic demodulation and beamformed coherence. Chenyang Zhu (Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, zhu.che@husky.neu.edu), Hamed Mohebbi-Kalkhoran, Sai Geetha Seri, and Purnima R. Makris (Northeastern Univ., Boston, MA)

Narrow-band tonal signals generated by ocean vessels are a dominant source of underwater sound, occurring at discrete frequencies ranging from several Hertz to several kiloHertz and continuously over long time intervals. Here we develop a two-stage approach for the automated passive acoustic detection and classification of underwater vehicles. The first stage utilizes the commonly used cyclic demodulation to extract the low frequency tonal sound modulated into high frequency noisy signal from which the fundamental frequency and blade pass frequency of ship propeller can be determined. The second applies magnitude-squared coherence to beamformed signals to directly extract tonal sounds spanning a wide range of frequencies from low ($\sim$10 Hz) up to the array cutoff frequency and to estimate their bearings. Both the modulated and unmodulated tonal signals generated by a ship can be simultaneously determined using this two-stage approach leading to a more complete spectral characterization of the underwater sound radiated by an ocean vessel. Here, the approach is applied to analyze and characterize the underwater sound generated by research vessel RV Knorr, received on a large-aperture densely populated coherent hydrophone array.

3pUWb3. Localizing underwater acoustic pingers using a streamlined method with an autonomous unmanned vehicle. Emma Carline (None, 110 Parkway Dr., Truro Heights, NS B6L 1N8, Canada, emma.carline@oceansonics.com) and Mark Wood (None, Truro Heights, NS, Canada)

Locating assets underwater is an important role of AUVs and Gliders. One instance is the recovery of black boxes from downed aircraft. All aircraft have an acoustic beacon attached to their flight data recorder that activates upon impact with water. Currently, to find the recorder a ship will search a large area towing a hydrophone while human operators carefully listen for the beacon's pings. This is very costly! After searching the entire area, an estimate of the beacon's location is made based on the loudness and spatial distribution of detections. Is this reliable? Is there enough time before the beacon's battery dies? This talk presents a streamlined approach for locating pingers on the seabed. Here, an AUV replaces the ship, two hydrophones replace the towfish, and a small processor containing an automatic ping detector and localization algorithms replace human operators. The method exploits the AUV backseat driver in order to form synthetic hydrophone arrays and to diminish the search area after a ping is detected. Not only does this streamline the AUV's route but also pinpoints the location of the blackbox from far away. The method will be introduced and results of field trials shown.

3pUWb4. Study of transmission loss model using the phase of the source. Dawoon Lee (Dept. of Energy & Resources Eng., Korea Maritime and Ocean Univ., Busan, South Korea) and Wookeen Chung (Dept. of Energy & Resources Eng., Korea Maritime and Ocean Univ., 727, Taejong-ro, Yeongdo-gu, Busan 49112, South Korea, wkchung@kmou.ac.kr)

MOE(Measure Of Effectiveness analysis) is needed to predict the sonar detection performance and maneuvering tactics in the ocean. To effect analysis, the detection probability is calculated by sonar characteristics and sound velocity distribution. The detection probability considering the sonar characteristics and the sound velocity distribution can be derived by using the FOM (figure of merit) and the transmission loss. Generally, the transmission loss used to calculate the detection probability is obtained by calculating the path of ray and the attention for the frequency of the source, sound velocity distribution. Although these models have advantages of being efficient and fast, it cannot consider the effect of constructive and destructive

interference for the source. In this study, the propagation phenomena for source were calculated by solving the two-dimensional acoustic wave equation to obtain transmission loss. For the computational efficiency, the proposed method uses the geometric spreading correction for the amplitude correction of the three dimensions. The calculated transmission loss is compared with transmission loss using BELLHOP, which is a sound transmission model. As a result, it is confirmed that the proposed method can consider the phase of the source.

3pUWb5. Calibration of marine differential pressure gauges through *in situ* testing at the seafloor. Adrian Doran (Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093, adoran@ucsd.edu), Martin Rapa, Gabi Laske, Jeff Babcock, and Sean McPeak (Scripps Inst. of Oceanogr., La Jolla, CA)

Differential pressure gauges (DPGs) are a standard component of modern broadband ocean-bottom seismometer instruments and have proven useful for observing a wide range of seismic and oceanographic phenomena. However, the response function of the DPG remains poorly known, limiting our ability to recover amplitude and phase information from seafloor pressure signals with high fidelity. The sensitivity and long-period response are difficult to calibrate in the lab, as they are known to vary with temperature and pressure and perhaps between sensors of the same design. We present the results of a field experiment designed to determine empirical response functions *in situ* by inducing a pre-defined pressure offset on a deployed instrument. The results compare favorably with calibrations estimated independently through post-deployment data analyses. Our study demonstrates that observed response functions can deviate from the nominal response by a factor of two or greater with regards to both the sensitivity and the time constant. Incorporating calibration devices such as those described here into future deployments may prove to be a cost-effective way to improve the accuracy and utility of differential pressure data.

3pUWb6. A low-cost, autonomous surface vehicle-based system for unexploded ordnance bistatic acoustic localization and classification. Erin Fischell (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., WHOI, MS 11, Woods Hole, MA 02543, efischell@whoi.edu), Kevin Manganini (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Daniel Plotnick (Dept. of Acoust., Univ. of Washington Appl. Phys. Lab., Seattle, WA)

A growing application for unmanned vehicle technology is the localization, classification and mitigation of underwater hazards such as munitions in shallow harbor environments. Because visual inspection of targets can be difficult or impossible in murky harbors and requires precise target localization, acoustic sensors such as sidescan sonar and synthetic aperture sonar (SAS) are used more extensively for munition detection missions. While these techniques can provide rich images of targets and the environment, postprocessing is often required and the sensors themselves are too expensive to be practical in multi-vehicle operations. A new system for low-cost, autonomous-surface-vehicle-based unexploded ordnance detection and localization is under development to provide a practical alternative to provide greater coverage via distributed, multi-vehicle missions. A mobile receiver array, fixed source geometry makes it possible to cover large areas while detecting man-made, aspect-dependent targets based on directional scattering from seabed targets. The system built on a Woods Hole Oceanographic Institution JetYak autonomous surface vehicle and results from experiments in summer 2019 in Ashumet Pond in Falmouth, Massachusetts are presented. [Sponsored by SERDP.]

3pUWb7. Modeling sound propagation in the Great Salt Lake. Gabriel H. Fronk (Phys., Brigham Young Univ., 1 Campus Dr., Provo, UT 84604, gherrickfronk@gmail.com) and Tracianne B. Neilsen (Phys., Brigham Young Univ., Provo, UT)

Acoustical waves can be monitored in a quiet underwater environment to identify noises, such as impacts and explosions. Our goal was to find the range of distance that acoustical sources can be detected in the Great Salt Lake. We measured the pressure from line explosives in the lake from varying distances to a triangle array of receivers. Various acoustical properties of the lake, like sound speed, density, and coefficient of attenuation, were then used in a range-independent propagation model to calculate transmission loss. Given the high salinity and shallow depth, the estimated transmission loss in the lake is large with this range-independent assumption. We found that in the best case scenario, there is a 60 dB re 1 μPa transmission loss at 3 km for most frequencies, which complicates source detection.

3pUWb8. Support vector machine for underwater acoustic signal range and environment classification. Stephanie Herron (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, sherron2@byu.edu), Traci Neilsen, David F. Van Komen (Phys., Brigham Young Univ., Provo, UT), David P. Knobles (KSA LLC, Austin, TX), Peter H. Dahl, and David R. Dall'Osto (Appl. Phys., Univ. of Washington, Seattle, WA)

While machine learning has become increasingly popular as a means to learn information from large datasets, the question remains how different machine learning models can best be used to improve source ranging and environmental classification. In the current research, machine learning is used to predict the distance and depth of an impulsive source and the seabed type given a set of acoustic signals in a shallow-water ocean environment. Multiple machine learning models have been developed for this problem, including a support vector machine classifier. This support vector machine was trained on synthetic datasets of varying sizes and characteristics to predict a class corresponding to source-receiver range and seabed type. The trained model was able to classify synthetic acoustic signals with eighty-five to ninety percent average accuracy. The trained model is also applied to pressure time series signals recorded on the APL-UW Intensity Vector Autonomous Receiver system during SBCEX. Future work will continue to compare the efficiency and accuracy of support vector machines against other types of machine learning in underwater acoustics. [Work supported by the Office of Naval Research.]

3pUWb9. Research on shallow sea multiple ocean bottom seismometer positioning with GNSS/acoustic technique and echo sounder. Huimin Liu (School of GeoSci., China Univ. of Petroleum (East China), Changjiang west Rd., Qingdao 266580, China, upcliuhm@foxmail.com) and Zhenjie Wang (School of GeoSci., China Univ. of Petroleum (East China), Qingdao, China)

Abstract: The Global Navigation Satellite System (GNSS) combined with acoustic technique has achieved significant economic benefits in oil exploration. During positioning of ocean bottom seismometers, hundreds of underwater transponders attached to seismometers are typically deployed. In order to improve the positioning accuracy of multi-sensor in shallow water, this paper analyses the ship GNSS data, Long baseline system (LBL) and single-beam bathymetric data, and presents the solution method of multi-target underwater acoustic positioning in shallow water with the depth constraints. Combined with the Natural Neighbor Interpolation (NNI) method, the regional elevation model is constructed, and the transponders elevation interpolation is performed in the horizontal position of robust estimation solution. According to the symmetry observation and the smooth terrain of the construction sea area, the bending error of the sound line is further eliminated by using the new piecewise incident model of the sound line bending to improve the plane positioning accuracy of the multi-target underwater in shallow water. Experiments in the South China Sea show that the new method model is more stable, has better anti-gross error performance, and effectively improves the accuracy of positioning under the condition of inaccurate sound velocity measurements and more gross errors of observation data.

3pUWb10. Automated machine learning approaches for humpback whale vocalization classification. Hamed Mohebbi-Kalkhoran (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, mohebbikalkhoran.h@northeastern.edu), Chenyang Zhu, and Purnima Ratilal (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

The vocalization behavior of humpback whales was previously studied and mapped over instantaneous wide areas of the Gulf of Maine (GOM), spanning more than 100 km in diameter, using the passive ocean acoustic

waveguide remote sensing technique during their Fall feeding season. Acoustic signals were received on a 160-element hydrophone array system where beamforming was employed to significantly improve signal-to-noise ratio of the received vocalizations. The humpback whale vocalizations can be divided into two classes, song and non-song calls. Song vocalizations are composed of repeatable set of phrases with consistently short inter-pulse intervals. The non-song vocalizations, such as 'bow-shaped' and 'down-sweep' moans, have large and highly variable inter-pulse intervals and no repeatable pattern. Here we employ machine learning approaches to classify humpback whale vocalizations into song and non-song calls. Preprocessing methods including frequency filtering, wavelet denoising, beamforming, and spectral smoothing are applied. Several automated classification methods including Support Vector Machines, Gaussian Naive Bayes, and Neural Networks are explored. Implementation of these algorithms on the GOM dataset results in over 88% classification accuracy implying the machine learning approaches can be used in field studies for real-time classification.

3pUWb11. Validating passive acoustic methods for gas flux quantification, offshore Panarea, Mediterreanean Sea. Ben Roche (Ocean and Earth Sci., Univ. of Southampton, Flat 15, Rowan House, Hulse Rd., Southampton SO15 2SB, United Kingdom, Br4g13@soton.ac.uk), Jianghui Li (I), Paul White (ISVR / Ocean and Earth Sci., Univ. of Southampton, Southampton, United Kingdom), Jonathan M. Bull, John W. Davis (Ocean and Earth Sci., Univ. of Southampton, Southampton, United Kingdom), Michele Deponte, Emiliano Gordini, and Diego Cotterle (National Inst. of Oceanogr. and Appl. Geophys., Sgonico, Italy)

With the expected large scale adoption of carbon capture storage, a growing importance has been placed on developing methods for the detection and quantification of underwater gas releases in order to provide long term monitoring systems and assurance. Passive acoustic methods have strong potential, as a bubble larger than a few micrometers escaping from sediment into the water column produces a distinct acoustic signature, characterised by the Minnaert frequency. Building on this principle, Leighton and White (2012) devised a method for inverting the acoustic signature of a gas seep to determine its flux rate. However, until now little has been done to validate that passive acoustic gas inversion provides estimates that are consistent with other techniques, i.e., physical and optical measurements, in the field. A specialist acoustic-optical seabed lander was designed and deployed at a series of natural CO_2 seep sites in offshore Panarea with the objective of comparing simultaneous flux estimates. Our results show that not only are acoustic estimates reliably consistent with all other methods but are also far more efficient.

3pUWb12. Model-aided acoustic environment estimation via data fusion for autonomous underwater vehicles. Oscar A. Viquez (Massachusetts Intitute of Technol., 77 Massachusetts Ave., Bldg. 5-204, Cambridge, MA 02139, oviquezr@mit.edu), EeShan C. Bhatt, Michael Novitzky, and Henrik Schmidt (Massachusetts Intitute of Technol., Cambridge, MA)

In autonomous underwater vehicle (AUV) operations, high-fidelity acoustic and environmental models are necessary for facilitating autonomous behaviors. Our current method uses a virtual ocean simulator driven by environmental empirical orthogonal functions, which describe a pre-determined set of sound speed perturbations, and their corresponding recombination coefficients, weights assigned to each function to recreate a specific sound speed profile that can be updated via acoustic communication. However, this approach is limited by its reliance on pre-processed theoretical models of ocean dynamics and sparse CTD sampling, which is especially concerning in rapidly varying environments. As the mission is underway, it is advantageous for the vehicle itself to develop a better estimate of said recombination coefficients by accounting for on-board acoustic and environmental sensor data. We present an initial framework for bounding the estimate of the sound speed profile recombination weights against acoustic propagation models, under the constraints of a band-limited collaborative acoustic network and the risk-averse nature of AUV operations. [Work supported by the Office of Naval Research.]

3pUWb13. Acoustic characteristics of three bladderless fishes. Naizheng Yan (Graduate School of Fisheries Sci., Hokkaido Univ., 3-1-1, Minatocho, 音響研究室, 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), KOHEI HASEGAWA (Faculty of Fisheries Sci., Hokkaido Univ., Hakodate, Japan), and Naoyuki Kudo (Graduate School of Fisheries Sci., Hokkaido Univ., Hakodate, Japan)

The acoustic characteristics of bladderless fishes were examined by measuring the target strength (TS) of three species, pointhead flounder (flat-shaped), Arabesque greenling (spindle-shaped) and sandeel (cylinder-shaped). TS measurements were collected in a seawater tank ($10 \times 6 \times 5$ m) using the EK80 echo sounder (Simrad, Norway) and ES70-7C transducer (Simrad) over a frequency range of 45–90 kHz with a tether method. After TS measurement of the whole fish, we separated the flesh from the body (for pointhead flounder and arabesque greenling) and measured the TS of the head and bones. Pitch angle characteristics were measured from a head-down orientation (−30 deg) to a head-up orientation (30 deg), and TS was processed using Echoview 9 software (Echoview, Australia). TS patterns were examined with respect to orientation using the distorted-wave Born approximation model. The standard length, width, and height of fish were inputted into the model. The three bladderless fishes tested have different body shapes, meaning different areas are exposed to sound waves at the same body length. Therefore, the relationship between TS and the cross-section of the dorsal aspect of bladderless fish is discussed. [Work supported by the Sasakawa Scientific Research Grant from The Japan Science Society.]

3pUWb14. An information theory approach to assess acoustic-environmental significance. EeShan C. Bhatt (Mech. Eng., MIT, Rm. 5-223, 77 Massachusetts Ave., Cambridge, MA 02139, eesh@mit.edu), Oscar A. Viquez (Mech. Eng., MIT, Cambridge, MA), Michael Novitzky (CSAIL, MIT, Cambridge, MA), and Henrik Schmidt (Mech. Eng., MIT, Cambridge, MA)

Autonomous underwater vehicles (AUVs) rely heavily on the acoustic environment for tasks related to sensing, navigation, and communication. In advanced applications of these vehicles, such as those involving hazardous or highly dynamic regions, their internal representation of the oceanic environment can prove essential to the mission success. Our current method is a virtual ocean simulator described by environmental empirical orthogonal functions (EOFs), a pre-determined set of sound speed perturbations; and corresponding recombination coefficients, weights assigned to recreate a specific sound speed profile that can be updated via acoustic communication. Given the limited bandwidth of acoustic communications, it is necessary to transmit the most vital information to update the virtual ocean representation—the minimum necessary to most effectively capture the real-time variability. Here we present the framework for determining the acoustic significance and uncertainty of an environmental coefficient-EOF pair, using an upcoming experiment in the Beaufort Sea as a case study. [Work supported by the Office of Naval Research.]

Plenary Session and Awards Ceremony

Victor W. Sparrow,
President, Acoustical Society of America

Introduction of Recipients of ASA Scholarships

Presentation of Certificates to New Fellows

Jonas Braasch – For interdisciplinary contributions to musical acoustics and psychoacoustics of spatial audio technology

Qian-Jie Fu – For contributions to understanding auditory perception with cochlear implants

Chen-Fen Huang – For contributions to geoacoustic inversion for ocean state estimation

Yun Jing – For contributions to designing acoustic metamaterials and numerical modeling of wave propagation

Michael E. Ravicz – For contributions to measurements of middle-ear function

Martin D. Verweij – For contributions to nonlinear medical ultrasound

Presentation of Science Communication Awards

Quincy Whitney, journalist, for her book *American Luthier: Carleen Hutchins—The Art and Science of the Violin*

Dallas Taylor, Host and Executive Producer of Twenty Thousand Hertz podcast, and Kevin Edds, writer, producer, for the *episode 20,000 dBs Under the Sea*

Noel Hanna, ASA member and Leading Education Professional (Physics), UNSW Global, for his article *Explainer: Why the human voice is so versatile* in the online magazine *The Conversation*

Introduction of Award Recipients and Presentation of Awards

Medwin Prize in Acoustical Oceanography to Chen-Fen Huang

Rossing Prize in Acoustics Education to Preston S. Wilson

A. B. Wood Medal and Prize—Julien Bonnel

Silver Medal in Musical Acoustics to Murray Campbell

Silver Medal in Physical Acoustics to James M. Sabatier

The Hutchins Consort

The Hutchins Consort, led by Artistic Director Joe McNalley, plays a diverse repertoire on the violin octet designed and built by luthier Dr. Carleen Hutchins. These eight instruments, which range in size from the 18.5-inch treble to the 7.2-foot contrabass, are essentially scaled violins; each is designed to have a frequency response that is ideally suited to its playable range. Dr. Hutchins applied the process of free-plate tuning, which she pioneered, to create this family of instruments.

Based in Southern California, the Hutchins Consort brings together a group of extraordinary players to tackle the challenge of adapting the techniques of traditional strings, as well as inventing new techniques, to master the instruments Dr. Hutchins created. This ensemble of gifted musicians redefines the customarily accepted perception of the chamber music concert with programs that combine great works of virtually every musical genre-from the Renaissance to Rock.

Carleen Hutchins was the recipient of the first Silver Medal in Musical Acoustics "for outstanding contributions and leadership in the development of a new violin family of musical instruments, and for leadership in the acoustical research on bowed string musical instruments." She was awarded ASA Honorary Fellowship in 1998.

Quincy Whitney, who is receiving the Science Communication Award for her biography of Carleen Hutchins, will introduce the Consort. After the performance, members of the Consort will be available to answer questions from the audience.

WEDNESDAY AFTERNOON, 4 DECEMBER 2019 CAROUSEL, 5:30 P.M. TO 8:30 P.M.

Session 3eED

Education in Acoustics, Women in Acoustics, and Listen-Up and Get Involved!

Traci Neilsen, Cochair
Brigham Young Univ., Provo, Utah 84602

L. Keeta Jones, Cochair
Acoustical Society of America, 1305 Walt Whitman Rd., Suite 300, Melville, New York 11747

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, Pennsylvania 16801

This workshop for San Diego area Girl Scouts consists of hands-on tutorials, interactive demonstrations, and discussion about careers in acoustics. The primary goals of this workshop are to expose girls to opportunities in science and engineering and to interact with professionals in many areas of acoustics. A large number of volunteers are needed to make this a success. Please e-mail Keeta Jones (kjones@acousticalsociety.org) if you have time to help with either guiding the girls to the event and helping them get started (5:00 p.m. to 6:00 p.m.) or exploring principles and applications of acoustics with small groups (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

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics (4:45 p.m.) and Computational Acoustics (4:30 p.m.).
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 are as follows:

Engineering Acoustics (4:45 p.m.)	Stuart
Acoustical Oceanography	Empress
Animal Bioacoustics	Edison
Architectural Acoustics	Windsor
Physical Acoustics	Wilder
Psychological and Physiological Acoustics	Garden
Structural Acoustics and Vibration	Spreckels

Committees meeting on Wednesday are as follows:

Biomedical Acoustics	Hanover
Signal Processing in Acoustics	Empress

Committees meeting on Thursday are as follows:

Computational Acoustics (4:30 p.m.)	Stuart
Musical Acoustics	Coronet
Noise	Crystal/Continental
Speech Communication	Regent
Underwater Acoustics	Viceroy

THE INSTITUTE OF ACOUSTICS

A B WOOD MEDAL 2019

Julien Bonnel

A common requirement in ocean acoustic applications is for robust signal processing algorithms that take into account the complex ocean environment, which is variable in space and time. Julien Bonnel's research has provided the most innovative approach to signal processing of broadband acoustic data in the last decade: time and frequency warping.

Warping is a non-linear signal re-sampling method, based on a physical model of the shallow water sound channel, which compensates for dispersion in the propagating modes. The technique extracts the propagating modes from broadband signals in shallow water at significantly shorter ranges than previously possible, using data from only a single hydrophone.

The technique he has developed has applications in two important areas: geo-acoustic inversion in shallow water environments and passive detection and localization of marine mammals. An example of the latter is the localization based on whale vocalizations, to enable estimation of the animal's depth in the water. The method has been successfully applied to obtaining the range and depth of bowhead whales in the Chuchki Sea and North Atlantic Right Whales in the Bay of Fundy.

Julien continues his research in geo-acoustic inversion to develop methods for extracting modal wavenumbers from short horizontal arrays. Of significant general interest is his work on re-formulating the modal filtering technique for sound channels with different dispersion characteristics.

Julien Bonnel is a most promising early-career research scientist who has introduced innovative methods that have led to significant advances in ocean acoustics. The research community has benefitted from the use of his methods and he has done this within a few years of completing his PhD. Importantly, he continues to produce innovative research to improve the methods that he has already developed.

The IOA is delighted to present Julien Bonnel with the 2019 A B Wood Medal.

Silver Medal in Musical Acoustics

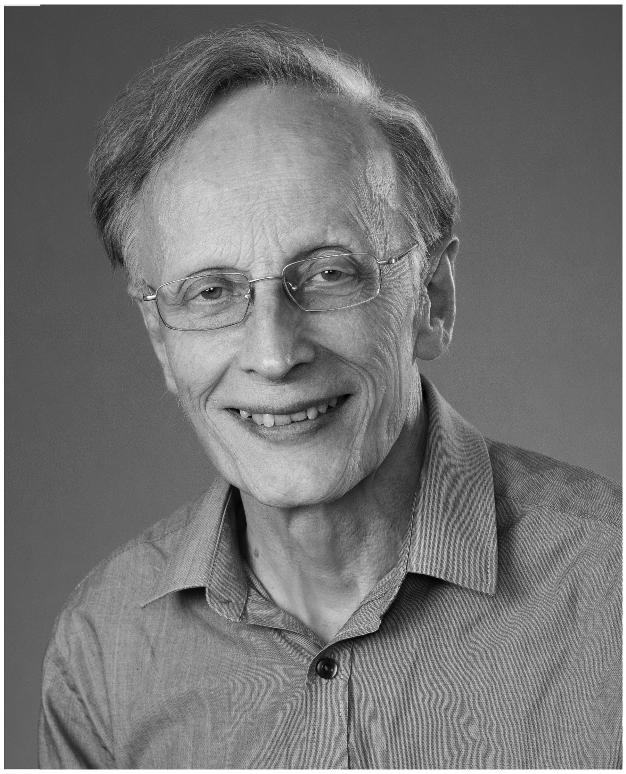
Murray Campbell

2019

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Carleen M. Hutchins	1981
Arthur H. Benade	1984
John G. Backus	1986
Max V. Mathews	1989
Thomas D. Rossing	1992
Neville H. Fletcher	1998
Johan E. F. Sundberg	2003
Gabriel Weinreich	2008
William J. Strong	2013



CITATION FOR MURRAY CAMPBELL

. . . for contributions to understanding the acoustics of brass wind instruments

SAN DIEGO, CALIFORNIA • 4 DECEMBER 2019

Like many musical acousticians, Murray Campbell did not begin his scientific career in acoustics. He was trained as an atomic physicist, but gravitated toward acoustics due to his interest in musical instruments. His extraordinary musical talent combined with his deep physical understanding made him a natural choice to teach a course in musical acoustics at the University of Edinburgh, which began a transition into the field where he would eventually be a driving force.

Professor Campbell received his BSc. and Ph.D. from the University of Edinburgh and was subsequently appointed Lecturer in Physics in 1972. He continued his work in atomic physics, but by 1985 Murray had also introduced a formal course in acoustics into the BSc. program and established the Musical Acoustic Research Group. This research group quickly became one of the preeminent collections of scientists and students performing research in musical acoustics, and his work eventually resulted in his appointment as Professor of Musical Acoustics.

In 1994 Murray published *The Musician's Guide to Acoustics* (Oxford Press), coauthored with Clive Greated, where his unique ability to connect acoustics with the details of musical performance in a language that both physicists and musicians can comprehend made it an international success. This book could only have been written by someone with Professor Campbell's musical talent and physical insight.

A decade later he published *Musical Instruments: History, Technology & Performance of Instruments of Western Music* (Oxford Press), coauthored with Clive Greated and Arnold Myers. This text is unique in its approach that combines clear descriptions of the science with explanations of the effects that each instrument had on the development of Western music. It is used throughout the world in introductory courses for students of both music and science. In addition to these texts, Professor Campbell is currently in the final stage of preparation of *The Science of Brass Instruments*, which will be published this year under the ASA/Springer imprint.

Professor Campbell's books have made a significant impact pedagogically and have secured his name as a public intellectual. However, his impact on the field of musical acoustics through the publication of his scientific articles cannot be overstated. His published works span the gamut of musical instruments. And many of his presentations involve him playing historic musical instruments with obvious skill, entertaining the audience with some humor while conveying a clear and scientific message.

Professor Campbell's contributions to the field of musical acoustics are many and diverse, but above all he is known as a pioneer in developing an understanding of brass wind instruments and the motion of the lip reed. Professor Campbell leads the field with his published works on these instruments and the nonlinear interactions that lead to their unique sound. His research on the interaction between the instrument and the player's lips is seminal, involving both theoretical and experimental work, and he has led the development of the study of "brassiness" in the sound of these instruments.

Although his professional accomplishments are many, Murray Campbell is probably best known in the community of acousticians as the consummate teacher and mentor. He has mentored numerous successful Ph.D. candidates and countless undergraduates since he began working in the field, and his former students now populate some of the most productive and prestigious research groups. However, Professor Campbell's influence on students extends far beyond those who have attended the University of Edinburgh. He always takes time at conferences to talk with students ranging from undergraduates to senior graduate students, freely offering ideas and support. He is arguably the most knowledgeable and caring musical acoustician in the world.

Dr. Campbell is an active musician, playing keyboard, string and brass instruments in numerous ensembles, as well as a symphony orchestra and jazz band. He is a Senior Honorary Professorial Fellow and Professor Emeritus of the University of Edinburgh, Fellow of the Acoustical Society of America, Fellow of the Royal Society of Edinburgh, and a Fellow of the Institute of Physics. He is also the musical director of the Edinburgh Renaissance Band and the Linton Singers, and the organist for his village church. He has been awarded the Rossing Prize by the ASA, the Médaille Etangére by the Societé Francais d'Acoustique, and the Stephens Medal by the Institute of Acoustics.

Murray Campbell is a world-renown acoustician, one of the most productive researchers in the field of musical acoustics, and has been a mentor to countless young scientists. His work is of the highest caliber, and his kindness and generosity are legendary in the community. Awarding Murray Campbell the Silver Medal is a deserved recognition of his many contributions to the advancement of our understanding of music and musical instruments, as well as his dedication to the development and education of the next generation of acousticians.

THOMAS R. MOORE
THOMAS D. ROSSING

Silver Medal in Physical Acoustics

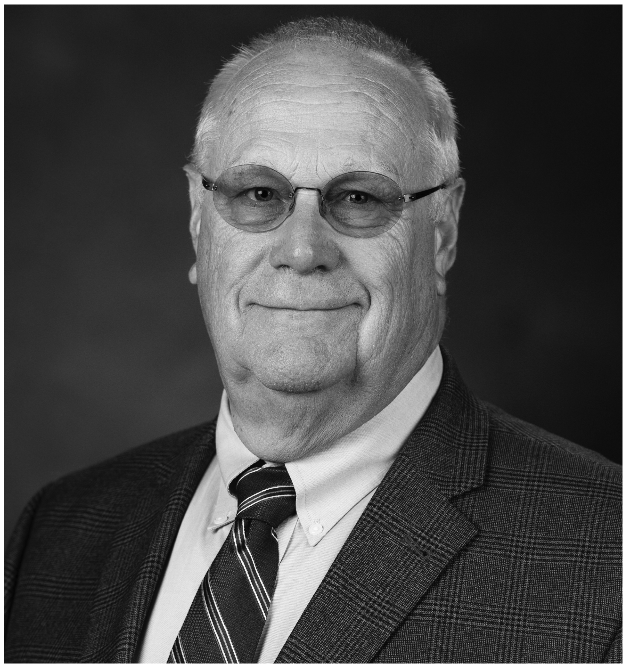
James M. Sabatier

2019

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Isadore Rudnick	1975	Robert E. Apfel	1997
Martin Greenspan	1977	Gregory W. Swift	2000
Herbert J. McSkimin	1979	Philip L. Marston	2003
David T. Blackstock	1985	Henry E. Bass	2006
Mack A. Breazeale	1988	Peter J. Westervelt	2008
Allan D. Pierce	1991	Andrea Prosperetti	2012
Julian D. Maynard	1994	Evgenia Zabolotskaya	2017



CITATION FOR JAMES M. SABATIER

. . . for pioneering studies of acoustic-seismic coupling and its application to the humanitarian cause of detecting landmines

SAN DIEGO, CALIFORNIA • 4 DECEMBER 2019

Jim Sabatier was born in Jennings, Louisiana. He earned a B.Sc. degree in Education at the University of Southwestern Louisiana in 1972 and joined Our Lady of Fatima School in Lafayette, Louisiana, where he served as Science Department Chair from 1972 to 1979. He then spent a year as Lecturer in the Department of Physics and Astronomy at the University of Southwest Louisiana before embarking on graduate studies at Ole Miss. As a graduate student, Jim worked on understanding and measuring acoustic-to-seismic coupling, the phenomenon by which airborne sound penetrates and, as a result of viscous drag at air-filled pore walls, causes soil-particle motion in porous ground. This resulted in the publication of three widely cited 1980s papers in the *Journal of the Acoustical Society of America* describing the theory of the physical mechanisms involved. He was awarded a Ph.D. in Physics by Ole Miss in 1984.

Since the pioneering work carried out for his Ph.D., Jim Sabatier has been at the forefront of research on acoustic-to-seismic coupling and its applications including acoustic monitoring of soil conditions and buried landmine detection. Conventional soil science techniques are laborious and invasive, often requiring sample extractions, and disrupting considerable proportions of the test plot surface. Jim showed the feasibility of relatively non-invasive acoustical techniques to replace these conventional methods. A succession of projects supported by USDA have shown the feasibility of using acoustic methods to determine the location of hardpans and to monitor soil erosion.

Having established that both linear and nonlinear soil particle motions induced by acoustic excitation above a buried landmine containing a cavity are anomalous, Jim carried out work to overcome the many engineering problems involved with using scanning Laser Doppler Vibrometry (LDV) from a moving platform to forward scan patterns of soil particle motion. Through extensive 'blind test' field trials supported by the US Army, Jim has shown the superior performance of the acoustic-seismic method over ground-penetrating radar. Although unclassified for many years, Jim's work on acoustic-seismic landmine detection is now classified. Jim achieved national notoriety by using an LDV system designed in connection with his landmine-detection studies to monitor the structural integrity of the partly collapsed World Trade Center tower building 4 to help with rescuing workers following the 9/11 disaster.

Jim's chosen areas of activity are inherently interdisciplinary. For example, his work on acoustical monitoring of soils encompasses acoustics, soil science and agricultural engineering. His interest in the development of practical technologies from his research means that he has frequently crossed the borders between physical and engineering acoustics. His refereed publication outlets include *IEEE Transactions*, *IEEE Geoscience and Remote Sensing Letters*, *Annali Di Geofisica*, *Radio Science*, *Optical Engineering*, *Reviews of Scientific Instruments*, the *Journal of the Acoustical Society of America*, and *Applied Acoustics*. In addition to his contributions to the Acoustical Society of America (ASA), Jim is a member of the International Society of Optical Engineering, where he serves as a committee member for symposia on landmine detection, and a member of the Soil Science Society of America.

During his career, Jim has held three Intergovernmental Personnel Act (IPA) Appointments since 2003 at the US Army Engineer Research and Development Center (ERDC) (1984), US Army Communications and Electronics Command (CECOM), Night Vision and Electronic Sensors Directorate (NVESD) (2001) and the US Army Communications and Electronics Command (CECOM), and the US Army Research Laboratory (ARL) (2009). He has served as a committee member for SPIE's Defense Security Sensing Symposium on the "Detection and Sensing of Mines, Explosive Objects and Obscured Targets," from 2003 to the present, as a US Army Red Team member for the Marine Corps Systems Command, Textron TRS SA Sensor (2006-2007), and as an Integrated Product Team member for the Joint Improvised Explosive Device Defeat Organization, 2005-2006. Since 1984, Jim has held a faculty position in the Department of Physics and Astronomy at Ole Miss.

During his third IPA appointment at the US Army Research Laboratory, Jim worked on personnel detection. The resulting research on ultrasonic characterization of human motion has resulted in a publication in the *Virtual Journal of Biological Physics Research*.

Two of his activities have resulted in patents. As well as contributing to the physical acoustics of sound-soil interaction and the ultrasonic signals associated with movement, his development of practical systems that apply this fundamental knowledge represent important contributions to engineering acoustics. Apart from contributions to soil-condition monitoring and landmine detection, Jim has used his ultrasonic-footfall signature work to devise a system for monitoring the mobility of elderly people.

His work on ground impedance has led to a version of a variable porosity ground impedance model, which enables better correlation with short-range propagation over grassland than many other models and, hence, is useful for outdoor noise predictions.

Jim's interest in and enthusiasm for science teaching has resulted in many presentations in schools, membership of the American Association of Physics Teachers (AAPT), and participation in Science road shows. He has presented demonstration sessions at ASA meetings in Norfolk VA and Nashville TN, the latter having drawn 150 middle and high school students, along with a large ASA audience. With the help of ASA volunteers, he presented a "Sounds of Science" demo session in Philadelphia for an AAPT meeting (2002). Jim has also had a strong commitment to the Physical Acoustics Summer Schools (PASS).

Jim was elected Chair of the ASA Education Committee in 2007. His work in the latter role from 2007-2010 resulted in many outstanding sessions at ASA meetings. In addition to the Education Committee, he has a long record of active participation in ASA Technical Committees (TCs) including as Chair of the Physical Acoustics TC from 1996-1999. Also he was the founding Chair of the Mid-South ASA Regional Chapter from 2007-2009.

Since 2012, Jim's mine detection research has been supported by the Office of Naval Research in support of advanced optical techniques to measure acoustic-to-seismic vibrations in addition to supporting the ground truth measurements and modeling. That work has led to development of a successful airborne optical and acoustic based platform for buried mine detection.

It is always fun to work with Jim Sabatier. Perhaps, a no better illustration of his experimental ingenuity and passion for acoustics, regardless of the potential risks to his well-being, is the story of one of his solutions to the problem of avoiding any seismic disturbance by direct contact during initial investigations of coupling of airborne sound with the ground. Jim is said to have jumped off a ladder and fired a starting pistol on the way down!

Thus, it is most fitting that Dr. James Sabatier be honored with the Silver Medal in Physical Acoustics of the Acoustical Society of America.

KEITH ATTENBOROUGH

Session 4aAA

Architectural Acoustics and Noise: Architectural Soundscapes I

Gary W. Siebein, Cochair
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Keely Siebein, Cochair
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Chair's Introduction—8:30

Invited Papers

8:35

4aAA1. Case study: Acoustic considerations, modelling, and auralization for large scale acoustic sculptures. Elizabeth Valmont (Acoust., Audiovisual and Theater, Arup, 900 Wilshire Boulevard, 19th Fl., Los Angeles, CA 90017, elizabeth.valmont@arup.com) and Brendan Smith (Acoust., Audiovisual and Theater, Arup, New York, NY)

This paper summarizes the research, measurements, observations, and modelling undertaken during the acoustic design of a sonic sculpture called Tower of Voices, part of the Flight 93 National Memorial in Stoystown, PA. The Tower of Voices is a 93 foot tall concrete tower featuring 40 large-scale wind chimes ranging in length from 5 to 10 feet. Due to the unprecedented nature of this project, the acoustic behavior of the chimes and interaction with the tower were the primary focus to achieve the goals of the client and designer for the soundscape visitor experience. Chime prototype mockups were undertaken to measure, record and observe sound propagation from the chimes, including the effects of different striker weights, damping materials, impact velocities, and strike locations. Acoustic modeling was utilized to assess the effect of tower shaping on the acoustic experience at various locations around the tower. An auralization was then prepared and presented in the Arup SoundLab to assist with design decisions regarding chime loudness and tower shaping. This project process of mockups, measurements, modelling, and auralization is described herein, along with a general recommended framework which can be applied to the acoustic design of large scale acoustic sculptures of many types.

8:55

4aAA2. Soundscape design using sound tube—To listen to voices, waves, wind, and footsteps. Hidemaro Shimoda (Acoust. Planning Corp., 9-6-29-309, Akasaka, Tokyo, Minato-ku 107-0052, Japan, shimoda@acoustic-plan.com) and Taiko Shono (Soundscape Lab., Tokyo, Shinjuku-ku, Japan)

In modern public architecture, a large amount of glass is often used as the material, and the space is visually connected by its transparency. However, on the other hand, the soundscape is divided. In the project of "Chino Cultural Complex," we intended to connect the two spaces through sound. This consists of a sound tube running approximately 20 m through underground of the courtyard. By speaking into one end of the tube, participants can converse with counterparts only faintly visible on the other side, with the voice as clear as if they were close to the ear. In this paper, we show the basic experiment of the sound tube in the anechoic chamber. We found an appropriate diameter of the tube in which the human voice was easy propagating on each other. Then, sound tube is covered with the absorbing materials and is set by the anti-vibrating suspension in the box culvert on site. The anti-vibration structure protected transmission of the structure-bone sound except voices to the sound tube. In other projects, we have designed this sound tube to listen to sound of waves, wind, and footsteps. We hope that these activities will provide significant soundscape design.

9:15

4aAA3. Acoustic sculpting revisited. Ganapathy Mahalingam (Architecture and Landscape Architecture, North Dakota State Univ., Dept. 2352, PO Box 6050, Fargo, ND 58108-6050, Ganapathy.Mahalingam@ndsu.edu)

The concept of acoustic sculpting was introduced in the early 1990s through a series of papers by the author. Acoustic sculpting is the process of generating the forms of spaces based on desired acoustical performance criteria for those spaces. These performance criteria are in the form of well known room acoustic parameters, such as the time-delay gap and clarity. This paper will revisit the methods of acoustic sculpting as described in the earlier papers and make efforts to strengthen the methods and expand the scope of acoustic sculpting. By critically analyzing room acoustical parameters and the information embedded in them, new ways to map that information into a process to generate spatial forms will be articulated. The methodology will be a process of mapping two different kinds of

information that are directly related, because the parameters of one are responsible for the generation of the parameters of the other. In traditional acoustical simulation, the spatial parameters of a space are known and the room acoustical parameters are derived through simulations. In the process of acoustic sculpting, this process is reversed, and the acoustical parameters are used to derive the spatial parameters. The purpose of developing the methods of acoustic sculpting is to use the mapping to create software systems for the generation of spatial forms for acoustical performance, such as the spatial forms of concert halls, auditoriums, theaters, and classrooms.

9:35

4aAA4. Performances immersed in or interacted with unique environmental contexts. Weihwa Chiang (Dept. of Architecture, National Taiwan Univ. of Sci. and Technol., 43 Keelung Rd. Section 4, Taipei 106, Taiwan, whch@mail.ntust.edu.tw)

By selecting the 100 soundscapes in 1996 and coordinating follow-up events in 2002, the Ministry of the Environment, Japan drew the attention to rediscover and preserve unique sounds across the country. The diverse soundcapes are associated with wildlife, natural phenomena, or man-made sounds from industrial or cultural activities. In addition to appreciating the sound events, performances at outdoor Noh stages in Japan were immersed in unique environmental context historically. Bird chirping, wind blowing, and sea tide may all become the context of the play. Similar situation could be found for historical plays of Chinese operas at the stage integrated with a Chinese garden. There are modern facilities or events designed to immersed in or interacted with natural or man-made context. Some take advantage of unique acoustic effect such as echoes in valleys while the others create focus by isolating ambient disturbances and integrating soundscapes with visual, functional, or other forms of environmental qualities.

9:55

4aAA5. The changing soundscape of the performing arts lobby. Jennifer Nelson Smid (Threshold Acoust., 141 W Jackson Blvd, Ste. 2080, Chicago, IL 60604, jnel04gt@gmail.com)

A lobby is traditionally the formal entrance to a performance venue. It can act as a buffer from the urban environment and be used by patrons before, during and after performances. In multi-venue performing arts centers, it can also act as buffer space between performances. An urban resurgence has seen younger populations move into cities and stay, and performance venues are adapting to serve the new populations. Performing artists have creatively extended their reach into the lobby using its spaces for lectures, pre and post performances, social gatherings and exhibits. Lobbies are becoming active during most of the day and evening, breaking down the acoustic buffer that they once held. This paper will focus on a soundscape study produced of several multi-performance venues that are finding creative ways to use their lobbies. It includes the identification of the activities that are taking place there, how these activities impact the main performance venues, and solutions to maintain the traditional functions of the lobby, whilst providing the active spaces needed for the future.

10:15–10:30 Break

10:30

4aAA6. The importance of ongoing observations in educational spaces to understand and reflect changing acoustical environments. Sang Bong Shin (Architecture, Univ. of Florida, UF School of Architecture, POB 115702, Gainesville, FL 32611, archisangbong@gmail.com)

In the past, observations to understand room acoustics were considered as an important tool and were performed to establish and develop acoustic experiments, parameters, and measurements. As the objective data have been collected, the standardized methods have been playing important roles in judgment and improvement of acoustical environments in educational spaces. These days, the standards for experiments or measurements have been commonly used to understand the quality of acoustic environments. Architectural soundscape has strongly suggested ongoing observations in order to reflect changing acoustic environments even though they show reliable and concrete data. For example, as the transition of educational environments has been made, the roles of teachers and students have been expanding to ones of each other. Also, the perceptions of sound have been rapidly changing since teachers' and students' diverse experience of industrial and electrical sonic environments have been changing rapidly. In this paper, observations on acoustic events in classrooms were taken to understand the changes in acoustical environments based on architectural soundscape in order how to change measurement settings or analysis methods. Also, acoustical measurements were conducted with diverse measurements setting including various positions of sound sources and receivers based on diverse educational arrangements shown in the results of the observations.

10:50

4aAA7. A study of classroom acoustics: Verification of perception responses according to time-variant sound exposure. Margret S. Engel (Sci. and Technol., Free Univ. Bozen-Bolzano, Franz-innerhofer-platz 5, Bozen, South Tirol 39100, Italy, margretsibylle.engel@unibz.it), Marco Caniato, and Andrea Gasparella (Sci. and Technol., Free Univ. Bozen-Bolzano, Bozen, South Tirol, Italy)

In educational contexts, learning processes focus on stimulation and development of students' cognitive abilities, which, in turn, will improve their learning capacity. Cognition refers to the mental activity of the process and uses information in judgment. Based on that, this study investigates how acoustic perception responses can differ according to sound exposure inside a classroom for different time spans. To achieve this aim, students and lecturers were asked to answer a survey about classroom comfort, which included acoustic perception questions, at a controlled environment classroom from the Free University Bozen-Bolzano. Perception responses could be given at any moment of the lecture, but it was recommended to answer the survey after the lecture. Additionally, two data collection campaigns were considered for this investigation: (1) without absorbers and (2) with fake absorbers. Sound pressure levels in octave bands were collected using a microphone array during the lectures period. The post-treatment of this data consisted of calculating sound pressure level averages for two different time spans: the whole lecture period and for the moment which the participants answered the survey. Room acoustic measurements were carried out to characterize the environment on different scenarios.

11:10

4aAA8. Longshan Temple soundscape. Lucky S. Tsaih (Dept. of Architecture, National Taiwan Univ. of Sci. and Technol., 43 Keelung Rd., Sec. 4, Taipei 10607, Taiwan, akustx@mail.ntust.edu.tw) and Yi-Ting Kuo (Dept. of Architecture, National Taiwan Univ. of Sci. and Technol., Taipei, Taiwan)

Longshan Temple, built in 1738, is one of the most visited historical religious places in Taipei. Longshan Temple was built with Chinese Siheyuan style, which is a traditional courtyard housing style. The objective of this soundscape pilot study is not only to identify and document the sound sources and paths in the temple but also to learn the perceived affective quality through the assessment of surrounding sound environment. The questionnaire used is suggested by ISO 12913. Binaural audio, video recording, and real-time sound pressure level were taken. The results show that sound from human being is dominated in the temple. The present surrounding sound environment is good with pleasant, eventful, and calm as the perceived qualities. With the measured equivalent sound pressure level at 68.9 dBA and "very loud" as the listening impression of temple, the soundscape of the temple is still rated with the pleasant and appropriated sound environment.

11:30

4aAA9. Urban spectralisms: Relational design approaches for orchestrating and sharing space. Nadine M. Schütz (Acoust. and Cognit. Spaces, IRCAM, 22 rue de Chabrol, Paris 75010, France, nadineschuetz@echora.ch)

If the study and design of architectural soundscapes stick to Euclidian notions of space, they risk overhearing the heterogeneous nature of auditory spatial experience beyond location and distance. A rich range of new musical aesthetics has led to the summarizing notion of *spectralisms* in the 20th century, describing compositional approaches built on mutual acoustic and perceptual considerations. But the design potential inhering this approach goes far beyond the realm of music. Spectral-temporal structures are key to grasping the spatial richness of our sonic environment. In urban planning, however, these dimensions are only considered as impacting the sensations evaluated through the psychoacoustic model of auditory pleasantness referring to single sounds and thus neglected as a domain of spatial formation. This paper argues that a mindful application of perceptual mechanisms related to the 'inner' acoustic structure of sound opens up new approaches for orchestrating and sharing urban space more efficiently. Through concrete project examples—such as the *Garden of Reflections* for the La Défense plaza or the *Acoustic Niches* for the forecourt of the new courthouse in Paris—it introduces *spectral thinking* as a conceptual tool for understanding and designing different forms of acoustic co-presence, comfort, and relation in our cities.

Session 4aAB

Animal Bioacoustics, Signal Processing in Acoustics, Acoustical Oceanography, and Computational Acoustics: Applications of Machine Learning to Bioacoustics I

Kaitlin E. Frasier, Cochair
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Kaitlin Palmer, Cochair
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Marie A. Roch, Cochair
Dept. of Computer Science, San Diego State University, 5500 Campanile Dr., San Diego, California 92182

Chair's Introduction—8:00

Invited Papers

8:05

4aAB1. Machine learning challenges 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), Michael Goldsworthy (Comput. Sci., Virginia Tech, Blacksburg, VA), Ruihao Wang, and Liujun Zhang (Mech. Eng., Virginia Tech, Blacksburg, VA)

Many bat species thrive in complex natural environments where their biosonar pulses trigger echoes with complex, unpredictable waveforms. In most cases, it remains unknown how the bats obtain sensory information they need from such clutter echoes. Machine learning methods that can extract relationships from large data sets could have a transformative impact on these research challenges. In particular, these methods hold considerable potential for answering the following questions: (i) What is the nature of biosonar echoes from complex environments; (ii) how can low-level navigation tasks such as contour-following and passageway finding be accomplished; (iii) what are acoustic landmarks for navigation and habitat selection; (iv) which clues exist for identifying prey in clutter; and (v) how does biosonar-based guidance in dense vegetation work at the system level? Machine learning methods are well suited for identifying models of echoes from complex vegetation that can be used to create large synthetic data sets with a known ground truth. Similarly, they can be employed to analyze (e.g., cluster) large physical echo datasets and to discover features associated with specific biosonar sensing tasks. Hence, machine learning could lead to the discovery of novel signal features to enable successful operation in much more complex environments.

8:25

4aAB2. A deep convolutional neural network based classifier for passive acoustic monitoring of neotropical katydids. Shyam Kumar Madhusudhana (BioAcoust. Res. Program, Cornell Lab of Ornithology, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, shyamm@cornell.edu), Laurel B. Symes, and Holger Klinck (Bioacoust. Res. Program, Cornell Lab of Ornithology, Cornell Univ., Ithaca, NY)

Insects occupy a central position in terrestrial trophic webs, consuming large amounts of vegetation and being consumed by other fauna. Changes in insect communities can have widespread effects across an ecosystem. In addition, having relatively small dispersal distances and stereotyped calls, insects make for an excellent choice as a monitoring target. Here, passive acoustic monitoring is employed for studying katydids, a relative of crickets and grasshoppers that are often charismatic mimics of leaves and sticks. In a prior study, call repertoires of over 60 species of katydids were established using individuals captured on Barro Colorado Island, Panama. These recordings are used in training a deep convolutional neural network based classifier, for subsequent analyses of long-term field recordings that are currently being collected. Coarse imbalances in the number of captured individuals per species and in their calling rates resulted in an unbalanced dataset. Furthermore, differences in recording equipment and in the recording practices posed challenges to achieving a well-rounded dataset. The measures taken to address these impediments will be discussed along with the classifier design choices. The classifier yielded >98% mean-average-precision over 60 classes on an exclusive validation split. Results of its application on field recordings will be presented.

8:45

4aAB3. Addressing challenges in automated whistle extraction. Pina Gruden (School of Ocean and Earth Sci. and Technol. (SOEST), Res. Corp. of the Univ. of Hawaii (RCUH), 1680 East West Rd., POST 802, Honolulu, HI 96822, pgruden@hawaii.edu) and Paul R. White (Inst. of Sound and Vib. Res. (ISVR), Univ. of Southampton, Southampton, United Kingdom)

Automated extraction of dolphin whistles from field recordings presents a challenging problem, as typically these recordings are noisy, contain multiple overlapping whistles and interfering signals. This results in many spurious detections, partial extraction and fragmentation of the whistle contours, which could cause problems for applications such as whistle-based localization. This paper presents the probability hypothesis density (PHD) filter, a non-traditional whistle tracking method based on multi-target Bayesian framework. Two implementations of the PHD filter, one based on Gaussian mixtures and one based on particle methods, are adapted for this task and tested on a large real-world dataset consisting of six dolphin species. The proposed filters successfully track simultaneous whistles, and compare favourably to standard methods. Moreover, it is shown how the proposed methods can be used to enhance the localization efforts with linear towed arrays.

9:05

4aAB4. A geospatial model of the global ambient soundscape. Shane V. Lympany (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, shane.lympany@blueridgeresearch.com), Michael M. James, Alexandria R. Salton (Blue Ridge Res. and Consulting, Asheville, NC), Matthew F. Calton (Blue Ridge Res. and Consulting, Provo, UT), Kent L. Gee, Mark K. Transtrum, and Katrina Pedersen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Humans and nature form an intricately coupled system with the ambient soundscape: anthropogenic, biological, and geophysical sources produce the sounds that comprise the soundscape, and, in turn, the ambient sound level affects the behavior and well-being of humans and animals. To assess the impact of the soundscape on both humans and animals, it is necessary to understand how the ambient sound level varies in space and time. A model for the ambient sound level was developed based on an ensemble of machine learning algorithms, which were trained using more than one million hours of ambient acoustic measurements acquired at hundreds of geospatially diverse locations across the United States. The resulting model predicts the ambient sound level based on geospatial features such as nighttime lights, land cover, population, climate, topography, hydrology, and transportation. A database of geospatial features with worldwide coverage was created, and the model was applied to predict the time-varying ambient sound level across the entirety of Earth's land surface. Furthermore, the relative contributions of anthropogenic and natural sources to the soundscape were estimated by artificially changing the values of various geospatial features and reapplying the model. [Work funded by an Army SBIR.]

9:25

4aAB5. Use of spatial context to increase acoustic classification accuracy. Kaitlin Palmer (San Diego State Univ., Dept of Comput. Sci., 5500 Campanile Dr, San Diego, CA 92182-7720, kaitlin.palmer@sdsu.edu), Tyler A. Helble (Space and Naval Warfare Systems Command, System Ctr. Pacific, US Navy, San Diego, CA), Eva-Marie Nosal (Dept. of Ocean and Resources Eng., Univ. of Hawai'i at Mānoa, Honolulu, HI), Glenn Ierley (Marine Physical Lab., Scripps Inst. of Oceanogr., Houghton, MI), Erica Fleishman (Dept. of Fish, Wildlife and Conservation Biology, Colorado State Univ., Fort Collins, CO), Yu Shiu (Bioacoust. Res. Program, Cornell Lab of Ornithology, Ithaca, NY), Xiaobai Liu (San Diego State Univ., San Diego, CA), Douglas M. Gillespie (Sea Mammal Res. Unit, Scottish Oceans Inst., Univ. of St. Andrews, St. Andrews, Fife, United Kingdom), Danielle Cholewiak (Northeast Fisheries Sci. Ctr., National Oceanic and Atmospheric Administration, Woods Hole, MA), Holger Klinck (Bioacoust. Res. Program, Cornell Lab of Ornithology, Ithaca, NY), and Marie A. Roch (San Diego State Univ., San Diego, CA)

Making classification decisions on the basis of multiple calls from the same class (e.g., species) can reduce the classification error that is attributable to variance. In many cases, clustering acoustic signals from multiple classes on the basis of location information can allow for separation and simultaneous classification of those calls. Source location and call timing are the most common criteria for clustering calls into groups or tracks. However, in many studies, widely spaced arrays or internal clock drift render a large proportion of the detected calls non-localizable. We built flexible clustering systems that incorporate all time difference of arrival values from calls detected by two or more sensors. We developed three clustering algorithms and used Monte Carlo simulations to compare their clustering accuracy and classification error. The algorithms increased clustering accuracy and reduced classification error by 2%–20% relative to grouping by time (a common fusion strategy) or classification of individual calls. These results demonstrate the ability of imperfect spatial information to improve classification accuracy of vocally active species.

9:45

4aAB6. An interactive machine learning toolkit for classifying impulsive signals in passive acoustic recordings. Kaitlin E. Frasier (Scripps Inst. of Oceanogr., 8622 Kennel Way, La Jolla, CA 92037, kfrasier@ucsd.edu), Alba Solsona Berga, John Hildebrand, Rebecca Cohen, Sean M. Wiggins (Scripps Inst. of Oceanogr., La Jolla, CA), Lance Garrison (SEFSC, NOAA, Miami, FL), Danielle Cholewiak (NEFSC, NOAA, Woods Hole, MA), and Melissa Soldevilla (SEFSC, NOAA, Miami, FL)

A typical wide-bandwidth passive acoustic seafloor sensor can record tens of millions of impulsive signals produced by biological, anthropogenic, and physical sources each year. Sources include echolocating toothed whales, snapping shrimp, ship propeller cavitation, echosounders, and weather. The volume and variety of detections make manual classification by human analysts unmanageable without in-depth knowledge of the overall acoustic context of each monitoring location. We describe an interactive machine learning toolkit in development for efficiently detecting and classifying short, highly variable impulsive signals in large passive acoustic datasets. Modules include a configurable event detector, an unsupervised clustering module for identifying dominant signal categories, a deep learning unit for learning and applying event classes, and a graphical user interface for viewing, correcting and evaluating detectors and classifiers. These tools are discussed and illustrated in the context of efforts to extract distinctive features of toothed whale clicks from towed array recordings collected by NOAA SEFSC and to use them to classify detections in unlabeled seafloor sensor recordings. The goal of the toolkit is to facilitate classification of individual signals to species in very large PAM datasets across sensor types and monitoring locations, and to improve quantitative assessment of these sources.

10:05

4aAB7. BirdVox: From flight call classification to full-season migration monitoring. Vincent Lostanlen (Cornell Lab of Ornithology, 370 Jay St., 13th Fl., New York, NY 11231, vincent.lostanlen@nyu.edu)

The BirdVox project aims at inventing new machine listening methods for the bioacoustic analysis of avian migration at the continental scale. It relies on an acoustic sensor network of low-cost, autonomous recording units to detect nocturnal flight calls and classify them in terms of family, genus, and species. As a result, each sensor produces a daily checklist of the species currently aloft, next to their respective individual counts. In this talk, I describe the research methods of BirdVox and their implications for advancing the understanding of animal behavior and conservation biology. The commonality of these methods is that they tightly integrate data-driven components alongside the induction of domain-specific knowledge. Furthermore, the resort to machine learning is not restricted to supervised acoustic event classification tasks, but also encompasses audio representation learning, few-shot active learning for efficient annotation, and Bayesian inference for adapting to multiple acoustic environments. I conclude with an overview of some open-source software tools for large-scale bioacoustics: librosa (spectrogram analysis), pysox (audio transformations), JAMS (rich annotation of audio events), muda (data augmentation), scaper (soundscape synthesis), pescador (stochastic sampling), and mir_{eval} (evaluation).

10:25–10:40 Break

Contributed Papers

10:40

4aAB8. Fine-scale classification of odontocete echolocation clicks using deep learning. Macey Rafter (Scripps Inst. of Oceanogr., La Jolla, CA), Kaitlin E. Frasier (Scripps Inst. of Oceanogr., 8622 Kennel Way, La Jolla, CA 92037, kfrasier@ucsd.edu), Jennifer S. Trickey, Sean M. Wiggins, and John Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA)

All toothed whale species produce echolocation clicks, which can be used for passive acoustic monitoring. However in biodiverse habitats, such as off the coast of southern California, it is common for acoustic encounters with multiple species to overlap in time. This creates a challenge for acoustic classification in the context of long term monitoring. Machine learning methods may facilitate classification at the level of individual detections. A generic echolocation click detector was applied to passive acoustic recordings from the 2013 DCL dataset, which has manual labels of five odontocete species encounters available as a ground truth. We evaluated two different methods for training deep neural networks to classify the detections to species, using either individual click waveforms or clustered sets of similar echolocation clicks. Results show that deep neural networks have the potential to accurately classify clicks to species at fine scales, allowing for improved handling of complex acoustic environments and multi-species encounters. We suggest that click-level classification can facilitate more advanced quantitative analysis of passive acoustic recordings, and that it can be done efficiently using deep learning.

10:55

4aAB9. Developing a click type "library" for Hawaiian odontocetes using machine learning methods. Morgan Ziegenhorn (Biological Oceanogr., Scripps Inst. of Oceanogr., 6182 Agee St., Unit 202, San Diego, CA 92122-3651, mziegenh@ucsd.edu), Kaitlin E. Frasier (Scripps Inst. of Oceanogr., La Jolla, CA), Erin M. Oleson, Jennifer L. Keating (Pacific Islands Fisheries Sci. Ctr., NOAA, Honolulu, HI), and Simone Baumann-Pickering (Scripps Inst. of Oceanogr., La Jolla, CA)

Though odontocete echolocation clicks are highly useful in passive acoustic monitoring, their acoustic features vary depending on behavior, orientation, and location relative to the receiver. Using unsupervised machine learning, regional sets of echolocation click types can be identified that allow for automated labeling of bioacoustic data while accounting for within-type signal variability. This methodology derives labels for clusters of click types based on parameters including spectral shape, inter-click interval, and click duration. Here we present classification results of echolocation clicks produced by Hawaiian odontocetes. Unsupervised methods were used to establish a set of click types for 10 years of recordings (200 kHz) collected from a bottom-mounted hydrophone off the coast of Kona, HI. These automated labels were compared with manual labels for species with established and recognizable clicks, which allowed for ground-truthing of some click types identified by the unsupervised machine learning method. Towed array data (500 kHz) was used to assign click types to species with visual verification, expanding the set of known species-specific click types for the region. The click types and method established here will be useful in future analyses of Hawaiian odontocete density estimation, abundance, and distribution patterns using acoustic data without requiring visual confirmation.

11:10

4aAB10. Automated classification of dolphin whistles based on the convolutional neural network. Gang Qiao (Harbin Eng. Univ., Harbin, China), Lei Li (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, Heilongjiang 150000, China, lileidahai@163.com), Songzuo Liu, and Xin Qing (Harbin Eng. Univ., Harbin, China)

A number of classification systems for dolphin whistles are applied to study the relation between dolphin whistles and their behaviors. Traditional approaches require prior knowledge, numerous pre-processing and manually extracting the features by conventional signal processing methods. In this study, deep convolutional neural networks are used to classify whistles and automatically learn the sound characteristics from training data with less pre-processing. The classification system is trained by using a database of 9 sorts of measured dolphin signals with nearly 4000 samples, and whistle contours in testing dataset are divided into six types, including constant frequency, upsweep, downsweep, concave, convex, and multiple. Finally, more than 90% for the classification accuracy rate is reached, and results show insensitivity to background noise. Therefore, the algorithm can be employed to study the potential relation between dolphin whistle signals and behaviors, and then facilitate future studies on dolphin habits.

11:25

4aAB11. Deep whistle contour: Recall-guided learning from synthesis. Pu Li (CSRC, San Diego State Univ., 5500 Campanile Dr., CSRC 555, San Diego, CA 92182, pli5270@sdsu.edu), Xiaobai Liu (CSRC, San Diego State Univ., San Diego, CA), Erica Fleishman (Dept. of Fish, Wildlife and Conservation Biology, Colorado State Univ., Fort Collins, CO), Kaitlin Palmer (CSRC, San Diego State Univ., San Diego, CA), Yu Shiu, Holger Klinck (Cornell Univ., Ithaca, NY), Tyler A. Helble (Spawar Systems Ctr., San Diego, CA), Douglas M. Gillespie (Univ. of St. Andrews, St. Andrews, Fife, United Kingdom), Eva-Marie Nosal (Univ. of Hawaii, Honolulu, HI), Danielle Cholewiak (National Oceanic and Atmospheric Administration, Woods Hole, MA), and Marie Roch (CSRC, San Diego State Univ., San Diego, CA)

This paper presents a learning based method for detecting whistles of toothed whales from underwater hydrophone recordings. Our method represents audio signals as time-frequency spectrogram and employs the Fully Convolution Network (FCN) to estimate for each spectrogram a map of

contour confidences that are used for extracting discrete whistle contours. To avoid the expensive efforts of annotating whistle contours, we develop a data synthesis approach to generate spectrogram-contour pairs using spectrogram of background environment and a small set of whistle contours. Our study suggests that the deep contour model can be effectively learned from these synthesized samples. However, it is costly and unnecessary to synthesize equal amount of samples for each spectrogram or contour. Instead, we present an alternative learning algorithm that synthesize samples only for those spectrogram or contours that are not well modeled by the current network, measured by recall rates of contour points for each spectrogram-contour sample. This recall-guided learning algorithm can adaptively synthesize difficult samples to boost learning effectiveness. We applied the proposed method to the public DCLDE2011 dataset to extract whistle contours. Results show that our method can improve state-of-the-art method up to 21.9% in terms of F-score for multiple odontocete species.

11:40

4aAB12. Using faster region-based convolutional neural network for automatic detection of baleen whale social calls. Jeppe H. Rasmussen (Dept. of Marine Biology, Texas A&M Univ. at Galveston, 200 Seawolf Parkway, Galveston, TX 77554, jeppehave@tamu.edu) and Ana Širović (Dept. of Marine Biology, Texas A&M Univ. at Galveston, Galveston, TX)

Using blue and fin whale calls for estimating population density of the two species from passive acoustic data is an active research topic. However, manually analyzing long-term data is extremely time consuming yet a necessary first step for such work. Blue and fin whale social calls are highly variable and quite similarity, which has resulted in challenges in developing automatic detectors for these calls. The applicability of faster region-based convolutional neural network (Faster R-CNN) method for speeding up this detection and classification task was explored. A large dataset of blue whale D calls and fin whale 40 Hz calls from southern California was used for training the network: 1378 spectrograms were created from 10-s sound clips each containing a call. The resulting images were contrast-enhanced and manually labeled with region of interest (ROI) before being applied as training images. The probability score for each ROI was modified to favor detections within previously measured frequency range of the calls. Testing shows this Faster R-CNN to have a very low miss- and false positive rate for both call types and is thus a highly promising tool for detecting and classifying these baleen whale social calls.

Session 4aAO

Acoustical Oceanography: Acoustic Propagation and Geoacoustic Inversion

Tarun K. Chandrayadula, Chair
Ocean Engineering, IIT Madras, 109 B Ocean Engineering, IIT Madras, Chennai, Tamil Nadu, 600036 India

Contributed Papers

8:15

4aAO1. Physical mechanisms of generation of the noise bursts associated with nonlinear internal tides in shallow water. Boris Katsnelson (Univ. of Haifa, 199 Abba Khouchy Ave., Haifa 3498838, Israel, bkatsnels@univ.haifa.ac.il), Oleg A. Godin (Naval Postgrad. School, Monterey, CA), and Qianchu Zhang (Univ. of Haifa, Haifa, Israel)

Recent paper of authors [*J. Acoust. Soc. Am.* **145**(3, Pt. 2), 1670 (2019)] reported observations of repeated increase in acoustic noise intensity during passage of trains of nonlinear internal gravity waves. Periodic increases of pressure fluctuations by up to 35–40 dB in the frequency band from 10 to 5000 Hz were recorded on multiple near-bottom hydrophones during Shallow Water 2006 experiment. The present paper extends the earlier analysis to a more diverse set of nonlinear internal wave events by exploiting the acoustic data obtained at sites with a wider range of water depths and over a longer observation period. Three distinct physical mechanisms are identified, which are responsible for increased noise in different frequency bands. The observed noise bursts are attributed to turbulence of the water flow past hydrophones, sediment transport, and sediment saltation. Correlation between the velocity of the internal tide-induced near-bottom currents and noise intensity at low and mid-frequencies is investigated. [Work supported by ONR, NSF, and BSF.]

8:30

4aAO2. Low-frequency sound propagation in short ranges: A case where leaky modes can be observed. Shima Abadi (Univ. of Washington, 18115 Campus Way NE, Box 358538, Bothell, WA 98011, abadi@uw.edu)

In an ocean waveguide, sound propagation can be modeled by the normal-modes theory. In a lossless ocean floor, the acoustic field is a finite sum of the normal modes that are trapped in the water column. These modes have real acoustic wave numbers and their equivalent rays are at angles less than the critical grazing angle. When attenuation is present in the ocean floor, the number of normal modes increases. Those additional modes, called "leaky modes," have complex acoustic wave numbers, and their equivalent rays are steeper than the critical grazing angle. Since leaky modes attenuate in short ranges, they are not typically observable in measurements. In this presentation, I show that leaky modes can be observed on the data collected during marine seismic reflection surveys. Marine seismic reflection surveys are used to image the structure of the seafloor. They generate low-frequency signals from airguns and record the reflections off the seafloor up to a few kilometers from the source. I use simulations and experiments to characterize the short-range propagation of airgun pulses in shallow and deep water. The experimental data utilized in this study are from the COAST experiment conducted in the Northeast Pacific continental shelf.

8:45

4aAO3. Philippine Sea observations of acoustic transmission phase and intensity fluctuations and comparisons to path-integral theory. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu), Bruce Cornuelle, Matthew Dzieciuch, Peter F. Worcester (Inst. of Geophys. and Planetary Phys., Scripps Inst. of Oceanogr., La Jolla, CA), and Tarun K. Chandrayadula (Ocean Eng., Indian Inst. of Technol. Madras, Chennai, Tamil Nadu, India)

From April 2010 to March 2011, low frequency acoustic transmissions were carried out on a 330-km radius pentagonal acoustic array with a sixth transceiver in the middle in the Philippine Sea. Herein, we describe the fluctuation statistics of the signals originating on the six transceivers and recorded on a water column spanning receiver array located slightly off center of the pentagon. The signals analyzed here are interpretable in terms of specific acoustic ray paths clearly separable in time. Acoustic field statistics treated include (1) variances of phase and intensity, (2) vertical coherence and intensity covariance, (3) glinting and fade out rates, and (4) intensity probability density functions. Several of these observables are compared to predictions from Feynman path integral theory assuming the Garrett-Munk internal wave model suitably modified using Philippine Sea oceanographic observations. Data and theory differ by at most a factor of 2 and reveal the wave propagation regimes of unsaturated, and partially saturated.

9:00

4aAO4. Observations of low frequency, long range propagation in the Philippine Sea and comparisons with mode transport theory. Tarun K. Chandrayadula (Ocean Eng., IIT Madras, 109 B, Chennai, Tamil Nadu 600036, India, tkchandr@iitm.ac.in), John A. Colosi (Oceanogr., Naval Postgrad. School, Monterey, CA), Peter F. Worcester, Matthew Dzieciuch (Univ. of California San Diego, La Jolla, CA), James Mercer, and Rex Andrew (Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

The year-long Philippine Sea (2010–2011) experiment was an extensive deep water propagation experiment in which there were six different sources transmitting to a water column spanning vertical line array. The six sources were placed across a 250 km $\times$ 250 km area, and transmitted at frequencies in the 200–300-Hz, and 140–205-Hz bandwidth. The PhilSea frequencies are higher than previous deep water experiments in the North Pacific for which modal analyses were performed. Further, the acoustic paths sample a two-dimensional area that is rich in internal tides, waves, and eddies. The PhilSea observations are thus a new opportunity to observe acoustic modal variability at higher frequencies than before and in an oceanographically dynamic region. This talk focuses on mode observations around the mid-water depths. The mode observations are used to present narrowband statistics such as transmission loss, and broadband statistics such as peak pulse intensity, travel time wander, time spreads, and scintillation indices. The observations are then compared with a new hybrid broadband transport theory. The model-data comparisons show excellent agreement for modes 1–10, and minor deviations for the rest. The discrepancies in the comparisons are related to limitations of the hybrid model, and other oceanographic fluctuations.

9:15

4aAO5. Arctic acoustic transmission loss variability due to ice cover during the year 2016–2017. Matthew Dzieciuch (SIO/UCSD, 9500 Gilman Dr., IGPP-0225, La Jolla, CA 92093-0225, mad@ucsd.edu) and Peter F. Worcester (SIO/UCSD, La Jolla, CA)

Over the course of one year, underwater acoustic transmissions were made in the deep-water of the Beaufort Sea as part of the Canada Basin Acoustic Propagation Experiment (CANAPE). The transmissions were monitored at a variety of ranges (100–300 km)and the low-frequency signals (200–300 Hz) showed a strong sensitivity to the evolving ice cover. As the ice cover grew from absent in the mid-September, to its thickest and roughest in late April, the transmission loss grew also. The experimental setup included the ability to beamform so that the arrival pattern could be easily resolved into early individual ray-paths and late near-surface arrivals. This capability was used to estimate the TL variation with the angle of interaction with the ice and with the number of ice reflections. The thickness of the ice cover and roughness was monitored at 6 mooring sites. This allows one to compare the measured losses to the expected loss from the ice-free state. The amount of excess loss can then be estimated via the method of small perturbations for example.

9:30

4aAO6. Spatial and temporal correlations of mid-frequency acoustic signals on the Chukchi Sea Shelf. Altan Turgut (Naval Res. Lab, Acoust. Div., Code 7160, Washington, DC 20375, altan.turgut@nrl.navy.mil), Michael D. Collins (Naval Res. Lab, Washington, DC), and Mohsen Badiey (Univ. of Delaware, Newark, DE)

The spatial and temporal correlations of 1.5–4 kHz LFM signals were measured using a 64-element billboard array and a mid-frequency acoustic source, deployed on the Chuckchi Sea Shelf during October 2016 to March 2017. The billboard array was deployed at 150 m water depth and the acoustic source was deployed at 320 m water depth near the shelf break to measure the acoustic signals propagating along a 20 km cross-shelf path. Both receiver depth (100 m) and source depth (200 m) were chosen to provide an effective transmission in a sound channel bounded by Pacific Summer Water at 40–100 m and warm Atlantic Water at 140-320 m below the surface. It was observed that signal amplitude variations and correlations were strongly affected by a combination of highly dynamic oceanography and Arctic sea ice. The effects of measured ocean and sea-ice variabilities were also simulated using a PE acoustic propagation model. The agreement between numerical predictions and measured data provides a framework for a robust mid-frequency sonar performance prediction capability in the Arctic. [Work supported by the ONR.]

9:45–10:00 Break

10:00

4aAO7. Out-of-band phase conjugation for auto-focusing and communications in the shallow ocean. YeonJoon Cheong (Mech. Eng., Univ. of Michigan, Ann Arbor, 2350 Hayward St., Ann Arbor, MI 48109, yjcheong@umich.edu) and Bogdan Ioan Popa (Mech. Eng., Univ. of Michigan, Ann Arbor, Ann Arbor, MI)

Time-reversal methods in which received sound is inverted in the time domain and retransmitted back towards the source allows effective temporal and spatial auto-focusing in complex and dynamically changing environments such as the ocean. However, time-reversal typically involves significant time lapses between the receipt of a signal and its time-reversed retransmission. Given the continuously changing nature of the ocean significantly affecting higher frequencies, demonstrations of time-reversal systems were limited to low frequencies below 5 kHz. The closely related phase conjugation methods remove the time gap between signal receipt and its retransmission but are difficult to implement because of additional hardware needed to decouple the receiver and transmitter operating on the same frequency band. Moreover, phase conjugation preserves only the spatial focusing of time-reversal and loses the temporal focusing ability. In this presentation we will introduce the concept of out-of-band phase conjugation in which the phase conjugated sound is on a higher harmonic of the received sound thus effectively decoupling the transmitter and receiver. At the same time, out-of-band phase conjugation improves the resolution of the auto-focused phase conjugated sound. We will also discuss methods to mitigate the effects of imperfect temporal focusing.

10:15

4aAO8. Improvements in signal processing for the estimation of environmental parameters with machine learning using a compact tetrahedral array and sources of opportunity. Jesse T. Moore (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Middleton 14, Narragansett, RI 02882, jesse_moore@my.uri.edu), James H Miller, Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Ying-Tsong Lin, Julien Bonnel, and Arthur Newhall (Woods Hole Oceanographic Inst., Woods Hole, MA)

In previous work, we showed that we could localize sound sources using 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 construction and operation of the first offshore wind farm in the United States. Directions of arrival (DOAs) for a number of ships were computed using a time difference of arrival technique. Given the DOAs, ranges are estimated using supervised machine learning techniques. We extended that work to estimate a number of environmental parameters including water depth and sediment composition. Here, we report on results using new spectrogram processing techniques based on high resolution PE modeling. These results include inversions for sediment parameters with estimates of error. With this new higher resolution spectrogram processing, we also report on the impact of the sediment parameters on range estimation. We generalize the technique to generic continental shelf environments. [Work supported by the Office of Naval Research.]

10:30

4aAO9. Estimation of geoacoustic parameters using machine learning techniques. Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg, Narragansett, RI 02882, potty@egr.uri.edu), James H Miller (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Zoi-Heleni Michalopoulou (Dept. of Mathematical Sci., New Jersey Inst. of Technol., Newark, NJ), and Julien Bonnel (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Modal dispersion of broadband acoustic data has been used extensively to estimate the geoacoustic parameters using perturbation techniques (Rajan *et al*., 1987), non-linear inversion techniques (Potty *et al*., 2000) and Bayesian inference (Warner *et al*., 2015). The Airy phase region corresponding the minimum group velocity of the normal modes are extremely sensitive to bottom properties in comparison with properties of the water column. This talk will explore the sensitivity of the group speed minima and the associated frequencies of the acoustic normal modes to bottom parameters. The group speed minima and the associated frequency data will be used to train a machine-learning algorithm. Synthetic data will be generated for various bottoms and it will be used as the training pool. Data collected from New England Bight, East China Sea, and New England Mud patch will be used to test the algorithm. The value added to the a priori bottom parameter information using the algorithm will be discussed in the context of computational cost associated with the algorithm. [Work supported by the Office of Naval Research.]

10:45

4aAO10. Investigating gradients in mud based on Bayesian modal-dispersion inversion and a hybrid geoacoustic model parameterization. Stan E. Dosso (School of Earth & Ocean Sci, Univ of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca), Julien Bonnel (Woods Hole Oceanographic Inst., Woods Hole, MA), N. Ross Chapman (School of Earth & Ocean Sci, Univ of Victoria, Victoria, BC, Canada), Preston S. Wilson (Univ. of Texas at Austin, Austin, TX), and David P. Knobles (KSA LLC, Austin, TX)

An important issue in understanding seabed geoacoustic properties at the site of the 2017 Seabed Characterization Experiment (SBCEX) on the New England Mud Patch is the extent to which gradients exist in geoacoustic properties (sound speed and density) over the upper mud layer. This paper applies Bayesian inversion and uncertainty quantification to modal dispersion data, resolved by warping analysis, to consider whether mud-layer gradients are required to fit the data. The inversion is based on a hybrid seabed-model parameterized in terms of an upper sediment layer with a general representation of smooth, continuous gradients in geoacoustic properties based on Bernstein-polynomial basis functions, above an unknown number of discrete layers formulated trans-dimensionally. The inversion results are compared to those from other acoustic data sets collected in the region as well as to nearby core measurements.

11:00

4aAO11. Trans-dimensional inversion of modal dispersion data collected by an underwater glider. Yong-Min Jiang (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC V8W2Y2, Canada, minj@uvic.ca), Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), Julien Bonnel (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and Preston S. Wilson (Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Acoustic payload equipped underwater gliders have been proven to have great potential for maritime intelligence, surveillance, and reconnaissance missions, as well as oceanic environment characterization. This paper demonstrates capabilities for seabed characterization using broadband signals received on a hydrophone-equipped Teledyne Webb Research Slocum glider during the 2017 Seabed Characterization Experiment (SBCEX) conducted on the New England Mud Patch. In the experiment, a source ship maintained a fixed position while two combustive sound-source signals were emitted at a separation of about two minutes in time. The glider was programmed to follow a sawtooth-like track through the water approximately 8 km from the source. Both source and glider were in an area where water depth is around 72 m. The two transmissions were received by the glider at depths separated by about 15 m. Discrepancies in the modal dispersion structure of the received signals were observed, as expected for different reception depths. Trans-dimensional geoacoustic inversion is applied to the modal-dispersion data to study the consistency of the inversion results for signals received at different depths. The benefit of combining signal receptions at different depths in the inversion to reduce the uncertainties of the geoacoustic estimates is also addressed. [Work supported by ONR.]

11:15

4aAO12. Broadband inversion in the seabed characterization experiment with a moving source. Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd, Newark, NJ 07102, michalop@njit.edu) and Peter Gerstoft (MPL, Scripps Inst. of Oceanogr., La Jolla, CA)

Sequential filtering and linearization for inversion of geometry and sediment parameter estimation in an oceanic environment can be achieved by employing particle filtering and linearization for a single source location. The inversion is here performed with linear frequency modulated signals received at vertically separated phones during the Seabed Characterization Experiment for a source moving towards and away from the receiving array. Short-range measurements allow the identification of some multipath arrivals, whereas others present us with a challenge. The filter, in combination with a simple cross-correlation, acts as a high-resolution estimator, allowing the extraction of four probability densities for arrival times. Arrival times are extracted for each source location. At every location, inversion is performed using the arrival time densities and both linearization and an exhaustive search relating unknown parameters to multiple paths. The results are validated by taking into consideration estimates at neighboring sites. We extract sediment sound speed and thickness assuming both a sound speed gradient and a sediment isovelocity profile and compare the results. [Work supported by ONR.]

Session 4aBAa

Biomedical Acoustics: Cavitation Bioeffects I

Hong Chen, Cochair
Washington University in St. Louis, 4511 Forest Park, St. Louis, Missouri 63108

Julianna Simon, Cochair
Graduate Program in Acoustics, The Pennsylvania State University, 201E Applied Science Building, University Park, Pennsylvania 16802

Chair's Introduction—8:00

Invited Paper

8:05

4aBAa1. Bioeffects elicited by acoustic cavitation. Christy K. Holland (Internal Medicine, Div. of Cardiovascular Health and Disease, and Biomedical Eng., Univ. of Cincinnati, Cardiovascular Ctr. Rm. 3935, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, Christy.Holland@uc.edu)

Ultrasound has been developed as both a diagnostic tool and a potent promoter of beneficial bioeffects for the treatment of cardiovascular disease and cancer. Ultrasound exposure can induce the release, delivery and enhanced efficacy of lytics, antibiotics, anti-inflammatory drugs or bioactive gases from echogenic liposomes. Other carriers for cancer therapeutics and small interfering ribonucleic acid (siRNA) and genes have also been developed. By encapsulating drugs and bubbles into micron-sized and nano-sized lipid-shelled particles, the therapeutic can be shielded from degradation within the vasculature until delivery is triggered by ultrasound exposure. The endothelial barrier between the bloodstream and vascular tissue presents a significant challenge to drug delivery. Most research on drug delivery from the blood vessel lumen into the tissue bed has been focused on blood-brain barrier disruption. Microbubbles oscillate when exposed to ultrasound and create stresses directly on nearby tissue or induce fluid effects that effect drug penetration into vascular tissue, lyse thrombi or direct drugs to optimal locations for delivery. Insonification accelerates clot breakdown in combination with a lytic and an ultrasound contrast agent, which nucleates sustained bubble activity. Mechanisms for ultrasound enhancement of clot dissolution, bactericide, and drug delivery, with a special emphasis on acoustic cavitation, radiation force and biological responses, will be reviewed.

Contributed Paper

8:25

4aBAa2. Contrast enhanced diagnostic ultrasound induces capillary injury in rat intestine. Douglas Miller (Radiology, Univ Michigan, 3240A Medical Sci. I, 1301 Katherine St., Ann Arbor, MI 48109-5667, douglm@umich.edu), Xiaofang Lu, Mario L. Fabiilli, and Chunyan Dou (Radiology, Univ Michigan, Ann Arbor, MI)

Contrast enhanced diagnostic ultrasound (CEDUS) can lead to microvascular injury through nucleation of cavitation. Capillary hemorrhage previously has been noted in heart, kidney, pancreas, and liver, which are common subjects of CEDUS examination. This research examined CEDUS injury of intestine. The abdomens of anesthetized rats were scanned by a 1.6 MHz diagnostic ultrasound probe during infusion of microbubble suspensions simulating the clinical ultrasound contrast agent Definity. Dual image frames were triggered intermittently, and the peak rarefactional pressure amplitude was varied to assess the exposure-response. Petechiae counts in small intestine mucosa and muscle layers increased with increasing trigger interval from 2 s to 10 s, indicative of a slow refill after microbubble destruction (cavitation nucleation). Counts increased with increasing output above a threshold of 1.4 MPa. Petechiae were also seen in Peyer's patches, and occult blood was detected in many affected segments of intestine. For comparison, thresholds were 0.73 MPa, 0.95 MPa, 1.1 MPa, and 1.7 MPa in kidney, heart, pancreas, and liver. Threshold variation may be related to differences in capillary size (largest in liver) and blood refill time (1 s in kidney). Clinically, such bioeffects can be mitigated by avoiding contrast agent destruction.

Invited Paper

8:40

4aBAa3. Drug delivery and treatment of neurodegenerative disease using focused ultrasound. Elisa Konofagou (Columbia Univ., 1210 Amsterdam Ave., ET351, New York, NY 10027, ek2191@columbia.edu)

Current treatments of neurological and neurodegenerative diseases are limited due to the lack of a truly non-invasive, transient, and regionally selective brain drug delivery method. The brain is particularly difficult to deliver drugs to because of the blood-brain barrier (BBB). The BBB prevents most neurologically active drugs from entering the brain and, as a result, has been isolated as the rate-limiting factor in brain drug delivery. Until a solution to the trans-BBB delivery problem is found, treatments of neurological diseases will remain impeded. Methods that combine Focused Ultrasound (FUS) and microbubbles have been shown to offer the unique capability of noninvasively, locally and transiently opening the BBB so as to treat central nervous system disease. Four of the main challenges that have been taken on by our group and discussed in this paper are (1) assess its safety profile, (2) unveil the mechanism by which the BBB opens and closes, (3) control and predict the opened BBB properties and duration of the opening, and (4) assess its potential in neurotherapeutics. Findings in both small (mice) and large (non-human primates) animals will be shown as well as its clinical translation for the treatment of neurodegenerative disease such as Alzheimer's and Parkinson's.

Contributed Paper

9:00

4aBAa4. Sonoprinting promotes drug delivery with liposome-loaded microbubbles and ultrasound. Michel Versluis (Phys. of Fluids group, Univ. of Twente, P.O. Box 217, Enschede 7500 AE, The Netherlands, m.versluis@utwente.nl), Guillaume Lajoinie (Phys. of Fluids group, Univ. of Twente, Enschede, The Netherlands), Silke Roovers, Joke Deprez, Ine Lentacker, and Stefaan De Smedt (Lab. of General Biochemistry and Physical Pharmacy, Ghent Res. Group on Nanomedicine, Ghent Univ., Gent, Belgium)

Ultrasound-triggered delivery from drug-loaded microbubbles has great potential due to its ability for localized release of the drugs while simultaneously enhancing delivery into the target tissue. We have recently proposed "sonoprinting" as a possible mechanism for the ultrasound-triggered delivery from nanoparticle-loaded microbubbles. It was shown that sonoprinting leads to the local deposition of nanoparticles and microbubble shell fragments in 2-D monolayer cultures. However, acoustic interactions between the bubbles and the rigid substrate could not be excluded. To verify whether sonoprinting can also occurs in more complex and physiologically more relevant tissues we study the US-triggered nanoparticle delivery from microbubbles to free-floating 3-D multicellular spheroids. Sonoprinting turned out to be a powerful tool to deliver large amounts of nanoparticles to the outer layers of tumor spheroids, followed by a complete drug release internalization into the deeper layers of the tumor spheroid. Sonoprinting significantly enhanced the cytotoxicity of both Doxil®-like and ThermoDOX®-like liposomes. We also provide evidence that only microbubble-associated nanoparticles become sonoprinted, while the uptake of free liposomal fraction is insignificant. As such, similar biological effects can be obtained through the use of substantially lower drug doses.

Invited Papers

9:15

4aBAa5. FUS-LBx: Focused ultrasound-enabled brain tumor liquid biopsy for noninvasive brain cancer diagnosis. Hong Chen (Washington Univ. in St. Louis, 630 West 168th St., PS 19-418, St. Louis, MO 63108, chenhongxjtu@gmail.com), Lifei Zhu, Christopher P. Pacia, Arash Nazeri, Yaoheng Yang, H. Michael Gach, Weijun Liu, Xiaowei Wang, Allegra Petti, Gavin Dunn, and Eric Leuthardt (Washington Univ. in St. Louis, St. Louis, MO)

Focused ultrasound (FUS) in combination with microbubbles has been studied extensively as a blood-brain barrier (BBB) disruption tool for the delivery of drugs in the blood circulation to the brain parenchyma. Recently, we introduced the hypothesis that FUS-mediated BBB disruption could be viewed as a tool for enhancing "two-way trafficking" between brain and blood. While circulating molecules can be allowed to enter the brain using FUS-mediated BBB disruption, brain tumor biomarkers can also be released into the blood circulation for liquid biopsies. Based on this hypothesis, we proposed to develop FUS-enabled brain tumor liquid biopsy technique (FUS-LBx), which uses FUS in combination with microbubbles to enhance the release of biomarkers from brain tumors into the blood circulation for liquid biopsies. We performed a proof-of-concept study and demonstrated the feasibility of FUS-LBx for the local release of mRNA from glioblastoma tumors in mice into the bloodstream for liquid biopsies. We also optimize the FUS-LBx technique by investigating the effects of FUS acoustic pressure on the tumor biomarker release level and potentially associated hemorrhage burden. We found that FUS-LBx technique can be optimized to be a safe and effective image-guided biomarker release technique. We developed an MR-compatible FUS system for FUS-LBx application in a porcine model and demonstrated the feasibility and safety of FUS-LBx in the large animal model. In summary, FUS-LBx is a promising tool for brain tumor diagnosis.

9:35

4aBAa6. Inertial cavitation behaviors and bioeffects in pulsed focused ultrasound. Tatiana D. Khokhlova (Harborview Medical Ctr., Univ. of Washington, 325 9th Ave., box 359634, Seattle, WA 98104, tdk7@uw.edu), Christopher Bawiec, Alex T. Peek (Univ. of Washington, Seattle, WA), Pavel Rosnitskiy (Faculty of Phys., M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Wayne Kreider, Adam D. Maxwell (Univ. of Washington, Seattle, WA), Vera A. Khokhlova (Faculty of Phys., M.V. Lomonosov Moscow State Univ., Seattle, WA), Helena Son, Yak-Nam Wang, Stephanie Totten (Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Faculty of Phys., M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), and Joo Ha Hwang (Dept. of Medicine, Stanford Univ., Seattle, WA)

Pulsed HIFU-induced inertial cavitation can be used to permeabilize tissue for promoting drug extravasation and penetration. In addition to HIFU frequency, peak rarefactional pressure (p^-), and pulse duration, the threshold for inducing inertial cavitation has recently been correlated with nonlinear distortion of the focal waveform. Here, the effect of nonlinear distortion and HIFU frequency within the range of 0.28–1.9 MHz were investigated in transparent gel phantoms with simultaneous high-speed photography and passive cavitation detection (PCD). Transducers with different F-numbers (0.77–1.5) delivered 1 ms-long pulses with varying p^- (1–15 MPa) at 1 Hz pulse rate. For all frequencies, increasing acoustic power and nonlinear distortion led to bubble behaviors changing both quantitatively and qualitatively from isolated, stationary bubbles to sparse bubble clouds that proliferated slowly towards the transducer. Different behaviors corresponded to specific spectral characteristics of the PCD signals. Exposures corresponding to particular behaviors at 1 MHz and 0.3 MHz frequencies were then applied to surgically exposed porcine liver, kidney, and pancreas *in vivo* with concurrent administration of Evans Blue Dye (EBD). Exposures with spectral signatures of proliferating bubbles on PCD corresponded to enhanced EBD extravasation and hemorrhagic injury to all tissues. [Work supported by NIH R01EB023910 and R01CA154451.]

9:55–10:10 Break

10:10

4aBAa7. Applications, mechanisms and real-time monitoring of enhanced drug transport by sustained inertial cavitation. Cameron Smith, Christophoros Mannaris, Prateek Katti, Catherine Paverd (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), Christian Coviello (OxSonics Ltd., Oxford, United Kingdom), Robert Carlisle (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom), and Constantin Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Old Rd. Campus Res. Bldg., Oxford OX3 7DQ, United Kingdom, constantin.coussios@eng.ox.ac.uk)

Sustained inertial cavitation seeded by microbubbles, sub-micron cavitation nuclei or even without any cavitation nucleation agent has been shown to enable increased delivery and penetration of drugs and vaccines across the skin, into solid tumours and across the blood-brain barrier. The safety and efficacy of these applications are typically monitored using single-element passive cavitation detectors, or more recently using multi-element arrays that enable real-time passive acoustic mapping of cavitation activity. We will provide an overview of the different types of cavitation nucleation agents which are available to use, the type and persistence of acoustic emissions that they are typically associated with, and, where possible, a direct comparison between the performance of different nucleation strategies to achieve the desired drug delivery bioeffect in the target of interest. We will then turn our attention to the presently available strategies for real-time monitoring of safety and efficacy, with specific reference to the benefits offered by the ability to achieve spatio-temporal monitoring of cavitation activity to correlate it with bioeffects in subsets of the overall region of interest.

Contributed Paper

10:30

4aBAa8. Defining a nuclei-independent unified cavitation dose for the spatiotemporal quantification of cavitation-mediated bioeffects by passive acoustic mapping. Cameron Smith (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford OX3 7DQ, United Kingdom, cameron.smith@magd.ox.ac.uk) and Constantin Coussios (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Cavitation can be seeded *in vivo* using a variety of nucleation agents with differing acoustic responses. The development of passive acoustic mapping (PAM) now makes it possible to spatio-temporally quantify the spectral content of acoustic emissions from cavitation. We present an approach that uses PAM to provide set-up independent measures of spatially dependent cavitation dose in units of Joules, and relate it to a readily quantifiable global bioeffect: the haemolysis of equine blood mixed with either micron-sized or sub-micron cavitation nuclei over a range of concentrations. Using two orthogonal L11-5 128-element arrays to achieve a 400-μm PAM resolution, the acoustic exposure parameters from a focused 0.5 MHz ultrasound transducer were varied from 0–2 MPa, and 50–50 000 cycles at 5% duty cycle to span a range of cavitation doses. A monotonic relationship between cavitation dose and haemolysis was identified, irrespective of the acoustic pressure, pulse length, cavitation agent or concentration used. The validity of this metric was tested further by using PAM to simultaneously monitor cavitation dose and haemolysis in multiple locations within a tissue phantom. These results lay the foundation for a unified and set-up independent cavitation dose metric for spatio-temporal monitoring of the safety and efficacy of cavitation-based therapies.

Invited Paper

10:45

4aBAa9. Multi-modal assessment of histotripsy liquefaction . Kenneth B. Bader (Dept. of Radiology, Univ. of Chicago, 5835 S. Cottage Grove Ave., MC 2026, Q301B, Chicago, IL 60637, baderk@uchicago.edu), Gregory Anthony (Graduate Program in Medical Phys., Univ. of Chicago, Chicago, IL), Viktor Bollen (Radiology, Univ. of Chicago, Chicago, IL), Samuel A. Hendley (Graduate Program in Medical Phys., Univ. of Chicago, Chicago, IL), and Steffen Sammet (Radiology, Univ. of Chicago, Chicago, IL)

Histotripsy is an emerging therapeutic ultrasound modality for mechanical tissue liquefaction via bubble cloud activity. Image guidance schemes for histotripsy necessitate both quantification of the bubble cloud activity and accurate delineation of the treatment zone. In this study, magnetic resonance (MR) and passive cavitation imaging (PCI) were combined to assess histotripsy treatment *in vitro* and *ex vivo*. Bubble cloud emissions were monitored with PCI. Changes in the medium structure were assessed with T_1-, T_2-, and diffusion-weighted MR images. Liquefaction zones generated in agarose phantoms and porcine livers were correlated with passive cavitation and MR images through receiver operating characteristic (ROC) analysis. Strong bubble activity was observed for all samples. Histotripsy-induced changes in sample structure were evident on T_2-weighted images for all samples, and T_1- and diffusion-weighted imaging for liver samples. The locations of the strongest MR pixel values did not necessarily coincide with the locations of the most intense bubble activity. The area under the ROC curve for predicting liquefaction was significantly greater than 0.5 for all imaging modalities, and PCI provided the best prediction of histotripsy damage. These results indicate PCI and MR imaging provide complementary sets of information, demonstrating the utility of multimodal imaging for monitoring of histotripsy.

Contributed Papers

11:05

4aBAa10. Real-time imaging and control of boiling histotripsy lesion formation using Doppler ultrasound. Vera A. Khokhlova (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., University of Washington, Seattle, USA; Phys. Faculty, Moscow State Univ., CIMU, APL, University of Washington, Seattle, WA 98105, verak2@uw.edu), Christopher R. Bawiec (Div. of Gastroenterology, Univ. of Washington School of Medicine, Seattle, WA), Bryan W. Cunitz (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., University of Washington, Seattle, USA; Phys. Faculty, Moscow State Univ., Seattle, WA), and Tatiana D. Khokhlova (Div. of Gastroenterology, Univ. of Washington School of Medicine, Seattle, WA)

In boiling histotripsy (BH), millisecond-long HIFU bursts with shocks are used to mechanically liquefy tissue. Most BH studies have been performed under real-time B-mode ultrasound imaging of bubbles generated at the treatment site. However, such approach does not allow quantitative assessment of the degree of tissue liquefaction. In addition, aberration and clutter often limit the echogenicity introduced by bubbles. In this work, plane wave Doppler followed by B-mode imaging were implemented immediately after each BH pulse to visualize residual bubble motion during volumetric BH treatments in *ex vivo* bovine liver. The changes in the Doppler power and speed were used as metrics to assess the degree of tissue liquefaction. An ATL P6-3 imaging probe connected to a Verasonics Ultrasound Engine (VUE) was mounted in the central opening of a 256-element 1.5 MHz HIFU array driven by another VUE with 1.2 kW external power source enhanced by a capacitor bank. The proposed imaging sequence improved BH targeting and bubble visualization. As tissue liquefaction progressed from partial to full (single versus five BH pulses per focal spot), both Doppler power and residual bubble velocity directed away from the transducer increased (7-fold and 10 cm/s to 20 cm/s, correspondingly) and then saturated. [Work supported by NIH R01EB7643.]

11:20

4aBAa11. Histotripsy in collagenous tendons. Molly Smallcomb (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, molly.smallcomb@gmail.com) and Julianna C. Simon (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Histotripsy is a high intensity focused ultrasound modality that uses cavitation bubble clouds or shock wave heating and millisecond boiling to mechanically fractionate most soft tissues. However, highly collagenous tissues like tendon have proven resistant to histotripsy fractionation. Previous work has shown that cavitation occurs during treatment but does not necessarily equate to detectable damage. In this study, collagen gels and *ex vivo* large animal Achilles tendons were exposed to focused ultrasound at 1.5 MHz, 1.07 MHz, or 3.68 MHz with 0.005–10 ms pulses repeated at 0.001–1 Hz for 15–60 s with acoustic pressures up to $p_+ = 88$ MPa, $p_- = 20$ MPa. A research ultrasound system was used to monitor for hyperechogenicity during the exposure. Tendon samples were collected and stained with hematoxylin and eosin (H&E) for analysis of cellular morphology and nicotinamide dinucleotide diaphorase (NADH-d) for analysis of enzymatic activity. Preliminary results show thermal denaturation rather than mechanical fractionation in tendon. Pilot studies in collagen gels show successful fractionation with histotripsy. Additional parameters will be evaluated on collagen gels and *ex vivo* tendon to promote mechanical fractionation rather than thermal denaturation. [Work supported by Penn State College of Engineering Seed Grant, NIH NIBIB EB027886, and NSF Graduate Research Fellowship DGE1255832.]

11:35

4aBAa12. Non-invasive treatment of abscesses by histotripsy. Yak-Nam Wang (Univ. of Washington, 1013 NE 40th St., box 355640, Seattle, WA 98105, ynwang@uw.edu), Andrew Brayman, Dan Leotta, Tatiana D. Khokhlova, Keith Chan, Wayne Monsky, and Thomas Matula (Univ. of Washington, Seattle, WA)

Abscesses are infected walled-off liquid collections of pus and bacteria. They can affect any part of the body. Current treatment is typically limited to antibiotics, catheter drainage and hospitalization. Although bacteria can develop drug resistance, they are susceptible to mechanical damage from cavitation, as demonstrated in our prior *in vitro* work [Brayman *et al.*, *UMB* **43**, 1476–1485 (2017); Brayman *et al.*, *UMB* **44**, 1996–2008 (2018)]. Histotripsy is a pulsed focused ultrasound regime that generates localized cavitation and represents a potential new noninvasive treatment modality. Here, we present preliminary reports of inactivation of bacteria in a newly developed porcine abscess model. In this model, multiple large (2–4 cm) multiloculated bimicrobial abscesses can be formed at distinct sites in the same animal. The abscesses satisfy the true definition of an abscess and are formed within a relatively short period (2–4 weeks). The model was developed toward testing new abscess treatment technologies. We will describe the animal model, and preliminary results of histotripsy treatments in which liquefaction of the viscous pus contents and up to 5-fold reduction in bacteria viability were achieved. Funded in part by NIH NIBIB #R01EB019365 and R01GM122859.

Session 4aBAb

Biomedical Acoustics: Biomedical Acoustics I

Yun Jing, Chair
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Contributed Papers

10:30

4aBAb1. Influence of compressive stresses on the ultrasonic response of the bone-implant interface. 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, Didier Geiger (Laboratoire MSME, Université Paris-Est, Créteil, France), and Guillaume Haiat (Laboratoire MSME, CNRS, Creteil, France)

The stress distribution around endosseous implants is an important determinant of the surgical success but it remains difficult to be measured. So far, no method developed to determine the implant stability is sensitive to the loading conditions of the bone-implant interface (BII). The objective of this study is to investigate whether a quantitative ultrasound (QUS) technique may be used to retrieve information on compressive stresses applied to the BII. A dedicated acousto-mechanical device was conceived to compress 18 trabecular bovine bone samples onto coin-shaped implants and to measure the ultrasonic response of the BII during compression. The biomechanical behavior of the trabecular bone samples was modeled as Neo-Hookean. The reflection coefficient of the BII was shown to decrease as a function of the compressive stress during the elastic compression of the trabecular bone samples and during the collapse of the trabecular network, with an average slope of -4.82 GPa^{-1}. Results may be explained by an increase of the bone-implant contact ratio and a change of bone structure occurring during compression. The sensitivity of the QUS response of the BII to compressive stresses opens new paths in the development of patient specific decision support systems allowing surgeons to assess implant stability.

10:45

4aBAb2. Ultrasonic guided wave propagation in a dental implant. 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), Bertrand Audoin (Institut de Mécanique et d'Ingénierie, CNRS, Talence, France), Chrstine Biateau (Institut de Mécanique et d'Ingénierie, CNRS, Bordeaux, France), Vu-Hieu Nguyen (Laboratoire MSME, Université Paris-Est, Creteil, France), and Guillaume Haiat (Laboratoire MSME, CNRS, Creteil, France)

Ultrasound techniques can be used to characterize and stimulate dental implant osseointegration. The acoustical energy transmitted to the bone-implant interface is an important parameter that must be controlled for both applications, since it should be sufficiently low to avoid damaging the surrounding tissues, but sufficiently high for stimulation purposes to enhance bone growth. However, the interaction between an ultrasonic wave and a dental implant remains unclear. The objective of this study combining experimental, analytical and numerical approaches is to investigate the propagation of an ultrasonic wave in a dental implant by assessing the amplitude of the displacements along the implant axis. An ultrasonic transducer was excited in transient regime at 10 MHz. Laser interferometric techniques were employed to measure the amplitude of the displacements, which varied between 5 and 12 nm according to the position. The results show the propagation of a guided wave mode along the implant axis with a velocity of first arriving signal equal to 2110 m s^{-1} and frequency components lower than 1 MHz, which was confirmed by the analytical and numerical results. This work paves the way to improve techniques for the characterization and stimulation of the bone-implant interface.

11:00

4aBAb3. Acoustic manipulation of large solid objects for medical applications. Mohamed A. Ghanem (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, 123529 4th Ave. ne, Seattle, WA 98125, mghanem@uw.edu), Adam D. Maxwell (Dept. of Urology, University of Washington School of Medicine, Univ. of Washington, Seattle, WA), Oleg A. Sapozhnikov (Ctr. of Industrial and Medical Ultrasound, Appl. Phys. Lab., University of Washington, Seattle, WA and Phys. Faculty, Moscow State Univ., Seattle, WA), and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Most progress in acoustic manipulation through radiation forces has been achieved in the trapping and levitation of small and low-density objects. We are developing traps to manipulate large dense objects. Such technology could be used to noninvasively steer a kidney stone along a complex path in the kidney fluid space to pass a small stone before it is symptomatic. In this work various acoustic beams were generated with a focused, 1.5-MHz, 256-element array and used to trap, levitate, and manipulate in three-dimensions (3-D) 3-5 mm glass spheres. The radiation forces from these beams were simulated and measured for different target and trap sizes. Good agreement between calculations and measurements of the forces was found, with an average discrepancy of 12 %. At 10-W acoustic power, the lateral trapping forces were 0.5–3 times the gravitational forces in water. 3-D stability and off-axis steering were achievable only when the ratio of beam diameter to the target diameter was $\gtrsim$1. Glass spheres placed in the focus of the array were manipulated over a preprogrammed 3-D path at pressure and intensity levels safe for clinical exposure. [Work supported by NIH P01-DK043881, K01-DK104854, R01-EB007643, and RBBR 17-02-00261.]

11:15

4aBAb4. Use sonography to assess swallowing in post-thyroidectomy patients. Yunying She (Otolaryngology-Head and Neck Surgery, Kaohsiung Veterans General Hospital, Taiwan (R.O.C.), No. 386, Dazhong 1st Rd., Zuoying Dist., Kaohsiung 81362, Taiwan, yyshe@vghks.gov.tw) and Hsinhua Chen (Otolaryngology-Head and Neck Surgery, Kaohsiung Veterans General Hospital, Taiwan (R.O.C.), Kaohsiung, Taiwan)

Every surgery which involved the patient's neck may have the chance to affect muscle kinetics related to functions of swallowing and voicing. Ultrasound can be a non-invasive diagnostic tool and provides the real-time image to visualize muscle kinetics. A prospective case control study. The aim of our study was to record the swallowing movements of patients after uncomplicated thyroidectomy postoperative 4–6 weeks and postoperatively 6 months using ultrasound . The patient will drink 10 ml water continuously with a repetitive saliva swallowing test. Analyzing the sonographic images to record the movement of cricoids cartilage while swallowing, the elevated

distance of the laryngeal box and the thickness of the belly of straps muscles to differentiated the inflammation status of muscles. Comparing these data with the normal cases can help us to define the diagnostic cut-off value for dysphagia and quantization the neck strangling objectively. The result revealed that the swelling of the strap muscles will affect the efficacy of laryngeal elevation and the motion of cricoid cartilage during the swallowing. This kind of examination for swallowing evaluation is convenient, quick, non-radiation, noninvasive, can help us to realize and focus on the real swallowing problem of the patient and improve the efficacy of the logopedic therapy.

11:30

4aBAb5. Spectral moment measures for the analysis of acoustic swallowing evaluation. Hsin-Hua Chen (Otorhinolaryngology - Head and Neck Surgery, Kaohsiung Veterans General Hospital, No. 386, Dazhong 1st Rd., Zuoying Dist., Kaohsiung City 81362, Taiwan, zoe.hhchen@gmail.com), Yunying She, and Yaoh-shiang Lin (Otorhinolaryngology - Head and Neck Surgery, Kaohsiung Veterans General Hospital, Kaohsiung City, Taiwan)

During pharyngeal phase of swallowing, related muscles work together to produce wave-like contractions and to move food into esophageal phase. How the peristaltic contraction of muscles and movement of bolus coordinate can significantly strengthen or weaken one's swallowing functions. This study uses the spectral moment analysis to investigate the acoustic effects of the peristaltic contraction wave during pharyngeal phase of swallowing. Four spectral moments describe features of the energy spectrum: the first moment (mean, centre of gravity) and the second moment (standard deviation) represent the spectrum's central tendency and dispersion respectively. Skewness and kurtosis (moment 3 and 4) represent spectrum's tilt and peak in relation to the central tendency. Results indicate that spectral moment measures may be one acoustic index sensitive to peristaltic contraction properties of swallowing. The statistical interpretation in accord with swallowing evaluation will be discussed.

11:45

4aBAb6. Automatic detection of involuntary spasticity of major muscles with acoustic and electromyography. Iyabo Lawal (Mech. Eng., William Marsh Rice Univ., 6100 Main St., Houston, TX 77005, igl2@rice.edu), Herbie Kirn (Articulate Labs, Austin, TX), Nicola Di Trani, Fernanda P. Pons-Faudoa, and Jesus Paez-Mayorga (Dept. of Nanomedicine, Houston Methodist Res. Inst., Houston, TX)

Spasticity is a severely debilitating condition where involuntary muscle contraction impedes normal movement, speech and gait. It usually results from motor root damage in patients with history of multiple sclerosis, traumatic brain injury, or stroke. This damage causes misfiring of neurons leading to erratic muscle contraction. Current treatments inhibit muscle activation, however, treatment is given upon onset of the spastic episode, delaying symptom alleviation. Here we developed a sensing device combining surface acoustic myography (sAMG) and surface electromyography (sEMG) for the automatic detection of spasticity. sAMG use acoustic-based sensors to detect pressure waves produced by muscle contraction whereas sEMG detect electrical signals from neuromuscular activity. Merging sEMG and sAMG sensors provided a robust set of spectral data to describe muscle contraction patterns in terms of electromechanical activities. Said patterns were used for calibration using machine-learning algorithms. With established normal patterns, the device can distinguish between voluntary and involuntary muscle movement. In sum, the symbiotic employment of both types of sensors shows potential in early detection of a spastic episode. Being able to detect the onset or to predict a spastic event can provide insight into the underlying physiological conditions and, ultimately, help develop technologies to mitigate its occurrence.

Session 4aEA

Engineering Acoustics: Transducers

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Contributed Papers

8:00

4aEA1. Experimental extraction of loudspeaker parameters from sound transmission measurements in a plane wave tube for moving-coil drivers and passive radiators. Brian E. Anderson (Dept. of Phys. & Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, bea@byu.edu), Joshua F. Gregg, Sarah M. Young, and Timothy W. Leishman (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

Moving-coil loudspeaker parameters are typically derived from electrical impedance measurements. Leishman and Anderson previously provided the theoretical foundations for extracting these parameters from sound transmission measurements conducted in a plane wave tube. This presentation will discuss experimental results of using this idea to extract the parameters of moving-coil drivers and those of passive radiators, the latter of which cannot be derived from electrical impedance measurements. The two-load method is used to account for the non-anechoic behavior of the tube to better enable accurate extraction of the sound transmission. Mechanical parameters are extracted from an open-circuit sound transmission measurement, and the electrical parameters are extracted by placing a wire across the loudspeaker's terminals to provide a closed-circuit sound transmission measurement. Parameters extracted from measurements of electrical impedance, sound transmission, and destructive analysis of diaphragm assemblies will be compared.

8:15

4aEA2. A heat balance model to explain delayed thermoacoustic sound production in seawater using metal electrodes. Michael S. McBeth (Naval Information Warfare Ctr. Atlantic, NIWC Atlantic/NASA Langley Res. Ctr., 8 North Dryden St., M.S. 473, Hampton, VA 23681, m.s.mcbeth@ieee.org)

Experiments conducted with solid metal wire electrodes in seawater generated second harmonic sound waves thought to be thermoacoustic in origin. Using sine wave bursts of five cycles of 10 kHz voltage across the electrodes, the thermoacoustic sound production was repeatedly observed to be delayed by about 240 μs from the initial applied voltage. At the time of these experiments around 2014, we were unable to account for this surprising delay in sound production. Recently, we built a heat balance model for the driven water volume that shows heat accumulating, during the applied voltage, faster than it can escape by heat conduction to the surrounding water. This results in a build up of heat in the driven water volume that increases the local temperature and density until a threshold level for sound production is reached. After the last cycle of the voltage burst, sound waves continue to be produced until enough heat has conducted away from the driven water volume for the temperature and density to fall below the sound production threshold. Although the model uses several simplifying assumptions, the results match the experimental observations reasonably well.

8:30

4aEA3. Pyroelectric ultrasound sensor model. Santeri J. Kaupinmäki (Dept. of Medical Phys. and Biomedical Eng., Univ. College London, Malet Pl. Eng. Bldg., Gower St., London WC1E 6BT, United Kingdom, jaakko.kaupinmaki.18@ucl.ac.uk), Ben Cox (Dept. of Medical Phys. and Biomedical Eng., Univ. College London, London, United Kingdom), Simon Arridge (Dept. of Comput. Sci., Univ. College London, London, United Kingdom), Christian Baker, David Sinden, and Bajram Zeqiri (Medical Phys. Dept., National Physical Lab., London, United Kingdom)

Ultrasound is typically measured using phase-sensitive piezoelectric sensors. Interest in phase-insensitive sensors has grown recently, with proposed applications in ultrasound attenuation tomography of the breast. The advantage of phase-insensitive imaging is that it does not suffer from degradation of image quality due to phase-aberration and narrow directional response. A numerical model of a phase-insensitive pyroelectric ultrasound sensor is presented. The model consists of three coupled components run in sequence: acoustic, thermal, and electrical. The acoustic simulation models the propagation and absorption of the incoming ultrasound wave. The negative divergence of the time-averaged acoustic intensity is used as a heat source in the thermal simulation for the time-evolution of temperature in the sensor. Both the acoustic and thermal simulations are performed using the k-Wave MATLAB toolbox with an assumption that shear waves are not supported in the medium. The final component of the model uses a pyroelectric circuit model which outputs the sensor response based on the temperature change in the sensor. The modelled pyroelectric sensor response and directional dependence are compared to empirical data. A physical model for the directionality of the sensor response is then proposed.

8:45

4aEA4. Modeling a piezoelectric cantilever beam with Simscape. Carter J. Childs (Graduate Program in Acoust., Penn State Univ., University Park, PA 16802, cjc357@psu.edu) and Stephen C. Thompson (Graduate Program in Acoust., Penn State Univ., University Park, PA)

Previous work developed a lumped parameter model in Simscape for the transverse vibration of a thin bar [C. J. Childs and S. C. Thompson, *J. Acoust. Soc. Am.* **145**, 1895 (2019)]. The current presentation expands that to include a layered bar, perhaps of different materials, and includes the electromechanical coupling for the case where the layers may be piezoelectric materials. This allows modeling such devices as the crystal or ceramic microphone that uses a piezoelectric bimorph element, and thin film piezoelectric cantilevers under consideration for vibration energy harvesting applications. While this paper uses Simscape for its examples, modeling these features in Modelica is equally possible.

9:00

4aEA5. A nonlinear lumped parameter model for designing a capacitive MEMS microphone. Chayeong Kim (Dept. of Mech. Eng., POSTECH, Hyoja-dong, Nam-gu, Pohang 790784, South Korea, kcycj@postech.ac.kr), Hongmin Ahn (Dept. of Mech. Eng., POSTECH, Pohang, Gyeongsangbuk-do, South Korea), Yoon Jong Lee, Han Choon Lee, Minhyun Jung (DB hitek Cooperation, Chungcheongbuk-do, South Korea), and Wonkyu Moon (Dept. of Mech. Eng., POSTECH, Pohang, Kyungbuk, South Korea)

The MEMS microphone, which is widely used for mobile devices, includes electro-mechanical energy conversion as well as acousto-mechanical conversion through the structures such as the back plate with many holes and an extremely flexible membrane with nonlinear characteristics. Hence, it is a time-consuming job to build an appropriate finite element model to predict the dynamic behaviors of the microphone. The equivalent circuit models are frequently used to check the linear behaviors of the MEMS microphone. However, it is sometimes inadequate to predict the nonlinear behaviors due to the nonlinear deformation of the sensing membrane, which usually determine the acoustic overload point. In this study, a nonlinear four-degree-of-freedom model is developed to design the MEMS microphone. Instead of the conventional equivalent circuit model, a model based on mechanical analysis is constructed and the state equations for the model is derived in the form of a set of ordinary differential equations. The responses of the model can be easily predicted in the time and frequency domain both by solving the equations numerically. With the time domain analysis used, the sensitivity can be easily shown as a function of frequency even including the nonlinear dynamic behaviors. [Work supported by DB Kim Jun-ki Cultural Foundation.]

9:15

4aEA6. Microelectromechanical systems directional acoustic sensor for underwater applications. Jason Roberts (Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jason.roberts@nps.edu), Alberto Espinoza, Fabio Alves, Renato Rabelo, and Gamani Karunasiri (Naval Postgrad. School, Monterey, CA)

In this work, a MEMS-based directional acoustic sensor operating underwater is explored. The sensor operates in a narrow frequency band centered at the mechanical resonance. The studied sensor consists of two wings coupled by a bridge which is pivoted to a substrate. Interdigitated comb finger capacitors attached to the wings allow for electronic readout of the mechanical oscillations, which are proportional to the sound direction of incidence. For underwater testing, the sensor was immersed in silicone oil, contained by a urethane housing with near unity acoustic transmission. The characteristics of the MEMS sensor both in air and silicone oil were analyzed using finite element modeling. Performance of the sensor was characterized both in air and underwater. Measured underwater frequency response showed that the resonance frequency of the sensor was shifted to a lower value compared to that of in air. This is primarily due to mass loading from the silicone oil used for immersing the sensor. Peak sensitivity of the sensor was found to be about 6 mV/Pa or –165 dB re 1 V/mPa. The sensor showed good directional response with a dipole pattern. Results show the potential of MEMS sensors for underwater applications to detect the bearing of sound sources.

9:30

4aEA7. Infrasound microphones using microfabricated piezoresistive silicon diaphragms. Randall P. Williams (Elec. and Comput. Eng., The Univ. of Texas at Austin, 10100 Burnet Rd., Bldg 160, Rm 1.108, Austin, TX 78758, randy.williams@utexas.edu), Yoonho Seo, Carly A. Stalder (Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), Bradley D. Avenson (Silicon Audio, Inc., Austin, TX), Quinlan S. Buoy (Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX), Donghwan Kim (Silicon Audio, Inc., Austin, TX), and Neal A. Hall (Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

We report on the development of a new digital infrasound measurement system with integrated digitizing electronics. The system was designed to meet an input-referred noise floor specification of 95 μPa/$\sqrt{}$Hz over the frequency range of 0.01 Hz to 100 Hz. The infrasound sensor uses a microfabricated silicon diaphragm with four integrated piezoresistors connected in a full-bridge configuration. Fabricating the sensing element on a silicon wafer using conventional micro-electromechanical systems (MEMS) processing techniques is a convenient approach for realizing sensitive diaphragms with integrated piezoresistors, and, in turn, sensing elements that intrinsically respond to DC. This approach is thought to be suitable for large infrasound arrays. Integrating a secondary transducer into the package provides self-test capabilities, along with a geophone for correlating infrasound signals with seismic vibrations. Sensor self-noise and sensitivity measurements are presented, along with a discussion of system design considerations, including design of the electronics and packaging.

9:45

4aEA8. Phase and amplitude correction of microphones for infrasound vector intensity using the noise field from a rocket motor. Francisco J. Irarrazabal (Phys. and Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, firarrazabal@gmail.com), Mylan R. Cook, Kent L. Gee, Scott D. Sommerfeldt (Phys. and Astron., Brigham Young Univ., Provo, UT), and Per Rasmussen (G.R.A.S. Sound & Vib. A/S, Holte, Denmark)

This paper describes the phase and amplitude correction of 12.7 mm diameter, Type-1 microphones in the infrasound region using the measured noise field of a large rocket motor. The data stem from acoustic intensity measurements using two-dimensional, three and four-microphone probes, which require that the acoustic phase and amplitude differences be much greater than the interchannel mismatch. Although processing for correcting the amplitude/phase is well-known, obtaining the necessary transfer functions in the infrasound regime is challenging because (1) signal-to-noise ratios are often poor, (2) long measurement times are required for averaging, and (3) microphone responses vary significantly across these low frequencies. Thus, a convenient noise source for performing a calibration is a static firing of a solid-fuel rocket booster. Far-field measurements of a large motor test greater than 1 min in length provide a known propagation direction and expected equivalence of significant, low-frequency amplitudes, which allows for correction of the microphone transfer functions at low frequencies. The performance of the calibration method, including limitations, for different microphones and probe configurations, is discussed. [Work supported by NSF.]

10:00–10:15 Break

10:15

4aEA9. Fly-inspired microphones with piezoelectric readout. Carly A. Stalder (Elec. and Comput. Eng., The Univ. of Texas at Austin, 10100 Burnet Rd., Bldg. 160, Austin, TX 78758, cstalder@utexas.edu), Donghwan Kim, Yoonho Seo, and Neal A. Hall (Elec. and Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

The directional hearing capabilities of the fly *Ormia Ochracea* have been known and utilized in MEMS devices since the 1990s. The fly locates crickets by listening to their chirps—an amazing feat when one considers the size of the fly's ear compared to the wavelength of audible sound. This work will focus on the design, modeling, fabrication, and testing of a biologically inspired microphone that harnesses this fly's hearing ability. Recent prototypes are 1/16th the size of previously fabricated devices of similar design and match the true size of the fly's hearing mechanism. The device consists of a two-sided cantilever beam that rotates about torsional pivots, resulting in two main frequency modes that can be used in the sound localization process for in-plane directivity: a gradient mode that causes the two sides of the beam to move out-of-phase, and a symmetrical mode that causes the two sides of the beam to move in-phase. The microphones are fabricated on a silicon-on-insulator wafer with a 2-μm-thick device layer. A 500-nm-thick aluminum nitride film is the piezoelectric transduction material. The main sensing modes are shown in multipoint scans and are accurately modeled with the finite element method.

10:30

4aEA10. Resolving bearing ambiguity with a single bio-inspired direction finding microelectromechanical system acoustic sensor. Brian Gureck (Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, brian.gureck@nps.edu), Renato Rabelo, Fabio Alves, and Gamani Karunasiri (Naval Postgrad. School, Monterey, CA)

Microelectromechanical systems (MEMS) directional sound sensors have been developed to mimic the unique hearing mechanism of the parasitoid fly *Ormia ochracea*. The fly analyzes the superposition of the two natural oscillation modes (rocking and bending) of its coupled eardrums to identify the incident direction of a cricket chirp. The incident sound generates unequal oscillation amplitudes of the eardrums depending on the direction of incidence. The MEMS sensor consisted of two wings connected by a bridge which is attached to a substrate using two torsional legs. For electronic readout of the oscillation amplitude of the wings, a comb finger capacitor was integrated to each wing. The amplitudes of oscillation of the two wings were independently measured to probe the coupling between the rocking and bending modes. The measurement of the sensor response was performed optically, using a scanning laser vibrometer, and electronically using a capacitance to voltage converter. It was found that the oscillation amplitudes of the two wings are strongly affected by the incident direction of sound. In addition, higher wing deflection occurred on the side closest to the incident sound source indicating the ability of the sensor to detect the direction of sound without the right-left ambiguity.

10:45

4aEA11. MEMS directional acoustic sensor with charge amplifier based electronic readout. Renato Rabelo (Dept. of Phys., Phys., Naval Postgrad. School, 833 Dyer Rd. - Spanagel Hall - Rm. 203, Monterey, CA 93943-5216, rcrabelo@nps.edu), Fabio Alves, and Gamani Karunasiri (Phys., Naval Postgrad. School, Monterey, CA)

MEMS acoustic sensors based on the operating principle of the *Ormia ochracea* fly provide a unique way to determine the direction of sound. A typical sensor consists of two wings coupled by a bridge and the entire mechanical structure attached to a substrate using two torsional legs. The sensor mechanical structure has two different vibration modes which can be coupled by placing their resonant frequencies close to each other. The acoustic wave impinging on the sensor induces displacements which are detected using comb finger capacitors at the edges of the wings. Previously, this has been accomplished using the universal capacitive readout MS3110, which is prone to static variations of the comb finger capacitors. In this work, a charge amplifier circuit was developed to independently measure the displacements of the wings under sound excitation. The ability to simultaneously read these displacements is needed to determine the bearing of sound using a single MEMS sensor, in contrast to previously reported works that needed either a calibrated microphone or a second MEMS sensor to determine the incident sound pressure level. Measurements showed that both the phase difference and amplitude difference of oscillating wings were strongly dependent on the incident angle of sound.

11:00

4aEA12. Liquid metal-based resistive membranes for flow acoustics detection. James P. Wissman (Acoust., NRC Postdoc at U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-0001, james.wissman.ctr@nrl.navy.mil), Kaushik Sampath (Acoust., NRC Postdoc at U.S. Naval Res. Lab., Washington, DC), Alec Ikei (Acoust., U.S. Naval Res. Lab., Washington, DC), Kadri B. Özütemiz, Majidi Carmel (Mech. Eng., Carnegie Mellon Univ., Pittsburgh, PA), and Charles Rohde (Acoust., U.S. Naval Res. Lab., Washington, DC)

Fluid flow over bodies results in vortex and turbulence-induced vibrations that reflect the boundary layer conditions and lead to drag and noise production. Many platforms would benefit from real-time monitoring of these acoustic signatures, providing feedback for control or increased efficiency. Of particular interest are compliant pressure sensors that can be wrapped around any surface. We present a silicone membrane integrated with a liquid phase gallium-indium coil. As the device deforms under pressure fluctuations, the resistance across the liquid metal can be monitored for real-time detection of flow features. The fabrication process is presented, including a masked spray deposition of gallium-indium coupled with layering of elastomer. The 18 mm-diameter sample demonstrates a point load sensitivity of 0.045 W/mm. To evaluate the performance of the device, it is tested in air with an acoustic tube, and its output is compared to impedance analysis at frequencies up to 10 kHz. Furthermore, the membrane vibrations are tracked with scanning laser Doppler vibrometer. These results are then compared to analytic models and COMSOL simulations. The sensor presented here is well suited to bulk fabrication in functional "skins" of active elements. [Work sponsored by the Office of Naval Research.]

11:15

4aEA13. Piezoelectric pressure sensors for hypersonic flow measurements at high-temperatures. Yoonho Seo (Elec. and Comput. Eng., UT Austin, Austin, TX, yhseo@utexas.edu), Donghwan Kim (Silicon Audio, Inc., Pflugerville, TX), and Neal A. Hall (Elec. and Comput. Eng., UT Austin, Austin, TX)

We present a microelectromechanical-systems (MEMS) piezoelectric pressure sensor designed for hypersonic flow measurements at extreme temperatures ($>$1000 °C). The sensing diaphragms have 700-μm diameter and are surface- and bulk-micromachined. The sensor employs aluminum nitride as a sensing layer due to its capability for high-temperature operation. We present the latest measurement results from a Ludwieg Mach-6 quiet tunnel, where prototype sensor data are compared against data from a state-of-the-art commercial sensor present in the same test. The prototype captures salient features of hypersonic flows including shock waves and second-mode instabilities indicative of laminar-to-turbulent transition. The prototype sensor also demonstrates superior signal-to-noise ratio in many test conditions. Prototypes have also demonstrated high-temperature operation by capturing audio while immersed in a butane flame ($\sim$1200 °C).

11:30

4aEA14. Ambiguity resolution using dual frequency transducer for acoustic Doppler current profiler applications. Sairajan Sarangapani (Electroacoust., Rowe Technologies, Inc., 12655, Danielson Ct, 306, Poway, CA 92064, ssairajan@gmail.com)

Broad Band Acoustic Doppler Current Profilers (ADCPs) have been used for over past 20 years to measure the relative velocity measurements in rivers and ocean environments but are inherently subjected to velocity ambiguities due to the phase change in the binary phase shift code. A new dual frequency transducer capable of transmitting and receiving at two different frequencies and having the same beam width is described here and is shown to resolve the inherent velocity ambiguity, thereby having velocity measurements with high accuracy and profile longer ranges from a single instrument. The electro-acoustic transducer made out of a piezoceramic in the shape of disc, vibrating in the extensional mode of operation is used in this approach to generate the directional acoustic radiation in the direction normal to the face of the transducer. The dual frequency transducer was integrated with the new generation ADCP-3 electronics, to transmit and receive sound underwater. An experimental investigation was conducted off San Diego coast using the dual frequency transducer operating at 2 different frequencies namely the low frequency at 300 kHz and high frequency at 1200 kHz in a single instrument and results are presented as a function of range and velocity measurements.

11:45

4aEA15. Preliminary analysis of operating conditions of acoustic sounder (SODAR) before the installation. Nishant Kumar (Acoust. and Vib. Metrology and Elec. and Instrumentation Eng. Dept., CSIR-National Physical Lab., New Delhi and Thapar Inst. of Eng. and Technol., Patiala, Dr. KS Krishnan Marg, Pusa, New Delhi, Delhi 110012, India, kumarnishant.kumar9@gmail.com), Kirti Soni (Acoust. and Vib. Metrology, CSIR-National Physical Lab., New Delhi, Delhi, India), Ravinder Agarwal (Elec. and Instrumentation Eng. Dept., Thapar Inst. of Eng. and Technol., Patiala, Punjab, India), and Mahavir Singh (Acoust. and Vib. Metrology, CSIR-National Physical Lab., New Delhi, Delhi, India)

The SOnic Detection And Ranging (SODAR) is a surface-based remote sensing instrument, based on Doppler effect and sound signal. This paper discusses the operating conditions of Monostatic SODAR

system, error in the measured Atmospheric Boundary Layer (ABL) height by taking account of the antenna characteristics which are pulse transmission, receiving durations, spectral augmentation of the received echo and atmospheric refraction effects. The Monostatic SODAR is able to receive even very weak echo by providing a good directional response with high conversion efficiency. The location and orientation of the antenna are finalized after a detailed analysis of the site with relative to several parameters like comparative measurement of acoustic background noise and frequency response of noise. Also, before installation of the system, the transducer has to be individually characterized for it's transmitted and received conversion efficiency, and variations in the product of these efficiencies (conversion gain). The antenna is systematically characterized, in an acoustic anechoic and reverberation chamber, with respect to its conversion efficiencies and directional response.

THURSDAY MORNING, 5 DECEMBER 2019 CORONET, 9:00 A.M. TO 11:15 A.M.

Session 4aMU

Musical Acoustics: General Topics in Musical Acoustics II

Andrew A. Piacsek, Chair
Physics, Central Washington University, 400 E. University Way, Ellensburg, Washington 98926

Contributed Papers

9:00

4aMU1. A comparison on sound quality of PLA 3-D printing Ukulele and single board wooden Ukulele. Xiaoyu Niu (Inst. of Acoust., Univ. of Chinese Acad. Sci., No. 21, North 4th Ring Rd., Beijing 100190, China, xiaoyuniu-ouc@foxmail.com), Peng Qian (Inst. of Acoust., Univ. of Chinese Acad. Sci., Beijing, China), Shanru Lin, Shumeng Yu (Ocean Technol., Ocean Univ. of China, Qingdao, Shandong, China), and Shengxue Fu (Ocean Univ. of China, Qingdao, Shandong, China)

With the development of science and technology in the 21st century, 3-D printing has made great progress. In the field of musical field, a complex sound source can be printed with the assistance of 3-D printing techniques. For example, the world's first Ukulele was printed out before long. Base on the innovation, we intend to compare and measure of the 3-D printing Ukulele and wooden Ukulele. Consequently, A-weight sound level of PLA 3-D printing Ukulele is less than that of traditional wooden with the same size, indicating that wooden resonant box is better at sound radiation than PLA 3-D printing Ukulele does. What's more, in the frequency domain, the fundamental frequency is higher than that of PLA 3-D printing Ukulele. In addition, in order to explain the difference between PLA 3-D printing Ukulele and wooden Ukulele, we simulate the vibrating state of two kinds of Ukulele with COMSOL. As a result, radiation acoustic impedance calculated by COMSOL could illustrate the difference. Hence, it is significant for musical instruments concerning the influence of material science.

9:15

4aMU2. Study of timbral influence of mallets on the steelpan through spectral analysis. Colin Malloy (Music, Univ. of Victoria, 3330 Richmond Rd., Victoria, BC V8P 4P1, Canada, malloyc@uvic.ca)

The steelpan is an instrument with unique acoustic properties that allow for a wide range of timbral possibilities influenced by the choice of actuator. It is increasingly common for composers and performers to experiment with mallets made from a variety of materials that produce highly differentiated timbres. Examples of mallet types include wood and aluminum shafts covered with rubber tips, chopsticks, dowel rods, and cardboard tubes. Understanding how mallets interact with the steelpan is an important aspect of performance practice. Aside from standard rubber tipped mallets, most other mallets types are homemade and these characterizations will inform mallet design and construction in order to achieve the desired timbral result. While there have been studies analyzing the modes of vibrations of steelpan notes, the interactions between mallets and timbre hasn't yet been studied and characterized. The goal of this work is to measure and characterize the timbral influence of a wide variety of mallets on high voiced steelpans through low level audio analysis with a focus on spectral features. Audio recordings were made with more than a dozen different mallets and several steelpans of three different types: tenor, double seconds, and double tenors.

9:30

4aMU3. From speech to songs via recital—A case of phase transition? A fractal study of Tagore's work. Archi Banerjee (Humanities and Social Sci., IIT Kharagpur, Kharagpur, West Bengal 721302, India, archibanerjee7@gmail.com) and Priyadarshi Patnaik (Humanities and Social Sci., IIT Kharagpur, Kharagpur, India)

This work explores the variation of scaling exponents in a song, recitation, and reading with the same lyrical content. Detrended Fluctuation Analysis (DFA) has been employed to find the long range temporal correlations (or the Hurst Exponent) present in each form of the auditory signal. Perceptually, it is known that the addition of rhythm and pitch, amplitude modulation distinguish between these different forms of audio signals, but the mathematical analogue of the same is still unknown. In this work, recordings were taken for 2 artists (1 male, 1 female) who were asked to read, recite and sing the entire lyrics of two (2) self chosen Tagore songs, each of which were later put to analysis. The rationale behind choosing Tagore's works is because of its strong lexical content. It was seen that Hurst Exponent is found to be the lowest in case of reading while it maximizes in case of song implying that the amount of long range correlation increases consistently with addition of rhythmic content and pitch, amplitude modulation in the audio signals. With this work, we tried to establish a critical value of Hurst exponent, above/below which transformation occurs to other forms analogous to critical temperature in phase transition of matter.

9:45

4aMU4. Modeling of wood resonances in prototype guitar top plates. Mitchel O'Conner (Phys., Whitman College, Walla Walla, WA) and Kurt Hoffman (Phys., Whitman College, 345 Boyer Ave., Hall of Sci., Walla Walla, WA 99362-2067, hoffman@whitman.edu)

We present the results of using COMSOL modeling tools to explore the resonance structure of prototype guitar top plates fashioned from several different grades of spruce. Comsol provides a platform for applying acoustical analysis to a variety of objects using the finite element method. Our studies utilize these techniques to explore the nature of normal mode vibrations in spruce top plates without bracing. The calculated normal modes will be compared to measured frequencies of prototype top plates. In addition, models of different grades of wood will also be explored in an effort to clarify which wood properties affect specific physical parameters for improved modeling.

10:00–10:15 Break

10:15

4aMU5. A platform for performance phase II: Evaluations of the structurally transmitted energy of the cello/performer system. David J. Tagg (Audio Eng. and Sound Production, Indiana Univ. Jacobs School of Music, 205 South Jordan Ave., Bloomington, IN 47405, jamtagg@indiana.edu)

Stringed instruments such as the cello have been evaluated extensively from structural and acoustical perspectives, but the riser on which a cello sits has until recently been overlooked. This riser becomes acoustically coupled and can play a productive or counterproductive role in its acoustic projection. The cellist's performance is understood to be enriched and enhanced by this piece of furniture, and this second installment in a two-part study further investigates the mechanical interaction between the musician, instrument, and supporting surface. This was accomplished by measuring the energy transmitted through five points of interface including the performer's heel, toe, front and rear legs of the chair, and endpin of the instrument. Results of the relative amplitude and frequency response of the structurally transmitted sound at these points, using five different performers, two performance techniques, and the full range of the instrument, will be presented.

10:30

4aMU6. Mode studies on note sections of a Gubal. Uwe J. Hansen (Phys., Utah Valley Univ., 64 Heritage Dr., Terre Haute, IN 47803-2374, uwe.hansen@indstate.edu), Joshua Gibson, and Dylan Morden

Over the years, Felix Rohner and Sabina Schärer, of Panart in Bern, Switzerland have implemented a number of significant modification to the standard Caribbean Steel Pan. These include using custom rolled sheet metal in place of commercially available used 55 Gallon drums, sinking the pan in a press rather than hand hammer sinking, dispensing with chiseled note section boundaries in favor of natural narrow change in curvature, adding central domes which give rise to more order in high overtone mode patterns, and building a lap held, hand played Hang. The latest addition to the Panart family is the Gubal, an inverted Hang with an added large dome on the bottom and the playing area, including the tone hole, at the top. Electronic Speckle Interferometric mode representations for Gubal note sections will be shown to illustrate spectral content of the Gubal.

10:45

4aMU7. Spectral analyses of the Ikoro drum. Stephen G. Onwubiko (Music, Univ. of Nigeria, Nsukka Enugu State, Enugu, Nsukka 234042, Nigeria, stephen.onwubiko@gmail.com) and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The Ikoro or Ekwe-is a wooden slit drum, a percussion instrument of the idiophone family which by virtue of its functions and its predominance in the eastern parts of Nigeria with the Igbo's plays an important role in the Igbo musical culture and settings. The Ekwe being a drum made of wood has pitches and produces a very deep rich bass sound which can be heard from a distance and has a sustained reverberation rendering a background beat to music or sound of warning. This paper looks at the frequency domain of the sustained reverberation, measuring, and analyzing the dynamics responses of the structure upon an excitation and describes the historical and constructional process of the Ekwe.

11:00

4aMU8. Mathematical music theory of embodied acoustics of Ikoro music using beat-class theory. Andrea Calilhanna (MARCS Inst. for Brain, Behaviour and Development, 2 Kayla Way, Cherrybrook, Sydney, New South Wales 2126, Australia, A.Calilhanna@westernsydney.edu.au) and Stephen G. Onwubiko (Music, Univ. of Nigeria, Enugu, Nsukka, Nigeria)

Through mathematical representation (beat-class theory) of embodied acoustics (psychoacoustics) the predominance of the musical tradition of the *Ikoro* drum with the Igbo's can be traced from the past, into the present and forecasted into the future. The *Ikoro* music tradition has been viewed as an integral and indispensable part of Igbo culture at large (Onwubiko and Neilsen, 2019). The major musical instruments that accompany most Igbo music are percussional, such as, *ichaka* (beaded-gourd rattle), *okpokolo* (wooden claves), and *igba* (membrane drum) and are characterized by successions of rhythmic interchange unlimited to interesting pitch, timbre, rhythm and meter by employing shifted accents, non-accented rhythms and syncopations. In order to understand *Ikoro* music located in the listener's experience (embodied psychoacoustics), we demonstrate how mathematical music theory (beat-class theory) provides the means to articulate the "mind and body" response to the stimulus of sound. By examining the aural tradition of *Ikoro* music of the Igbo's through visualizations and sonifications of beat-class theory using ski-hill graphs and circular cyclic graphs, "hidden" musical structures are revealed which possess significant cultural significance.

Session 4aNS

Noise, Physical Acoustics, and Computational Acoustics: Supersonic Jet Aeroacoustics I

Alan T. Wall, Cochair
Battlespace Acoustics Branch, Air Force Research Laboratory, Bldg. 441, Wright-Patterson AFB, Ohio 45324

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

Caroline P. Lubert, Cochair
Mathematics & Statistics, James Madison University, 301 Dixie Ave., Harrisonburg, Virginia 22807

Chair's Introduction—9:00

Contributed Papers

9:05

4aNS1. Experimental study on the acoustic field caused by the interference of supersonic jet with a perforated plate—The optical measurement of sound absorption. Yo Murata (Dept. of Mech. Eng., The Univ. of Tokyo, 7-3-1, Hongo, Bunkyo-ku, Tokyo 113-8656, Japan, ymurata@thml.t.u-tokyo.ac.jp), Kaku Murayama (Adv. Energy, The Univ. of Tokyo, Tokyo, Japan), Tatsuya Ishii (Japan Aerosp. Exploration Agency, Tokyo, Japan), Koji Okamoto (Adv. Energy, The Univ. of Tokyo, Kashiwa, Chiba, Japan), Hirofumi Daiguji, and SHIGEHIKO KANEKO (Dept. of Mech. Eng., The Univ. of Tokyo, Tokyo, Japan)

The supersonic jet is known to be a source to high intensity sounds. According to previous studies, when this jet collides with a perforated plate, a complex flow field is formed, and sound waves with much higher amplitudes are generated. In our study, an optical visualization test based on the Schlieren method was carried out, using a simple model consisting of a CD nozzle and a perforated plate. During the test, a sound-absorbing material was flush-mounted in the plate as an approach to alleviate the acoustic field. Visualized videos tried to account for absorption of high-amplitude sound waves by the sound-absorbing material. The videos were analyzed and compared with acoustic measurement for verification.

9:20

4aNS2. Entropy noise from shock containing nozzles of military jets at afterburner. Christopher Tam (Mathematics, Florida State Univ., 1017 Academic Way, Tallahassee, FL 323064510, tam@math.fsu.edu)

This paper considers the noise of military jets operating at afterburner. Boosting by the afterburner, the jet velocity and temperature are much higher than those of laboratory jets. Under these conditions, do we anticipate the noise components of these jets the same as those of laboratory jets, i.e., consisting mainly turbulent mixing noise? Do we expect new noise sources in these jets, since they operate at much higher velocities and temperatures and that the combustion processes in the afterburner are highly unsteady. In this paper, we explore one plausible new noise mechanism; the generation of entropy noise inside the shock containing nozzles of military jets. It is well-known that hot and cold spots (entropy waves) from unsteady combustion when convecting through a non-uniform mean flow would create fluctuating pressure and hence acoustic radiation. We investigate this possibility by performing numerical simulation of the flow and noise inside a rectangular nozzle typical of that of the F-22 Raptor. These nozzles are imperfectly expanded. There are shocks inside. We find that because shocks have very steep velocity gradients, their presence leads to the generation and emission of strong entropy noise when entropy waves from the afterburner pass through them.

9:35

4aNS3. Investigating focusing of Mach waves by Prandtl-Meyer expansion fan as an explanation for some spatiospectral lobe phenomena. S. Hales Swift (Energy Systems Div., Argonne National Lab., N221 ESC, Provo, UT 84602, hales.swift@gmail.com) and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Recent studies of high-performance aircraft have noted the presence of spatiospectral lobes: regions of physical and frequency space where the sound field exhibits increased amplitude not otherwise captured by typical jet noise models. This phenomenon results in multiple spectral peaks at particular positions and multiple spatial peaks at particular frequencies. These peaks have been variously explained as resulting from indirect combustion noise or shock-turbulence interactions of some type. This paper proposes to explain some aspects of the spatiospectral lobe phenomenon in terms of the focusing of Mach waves by the Prandtl-Meyer expansion fans that, with oblique shocks, comprise shock cells. Briefly, a higher convective velocity leads to an angle of propagation farther from the jet axis. Because the flow in an expansion fan is increasing in velocity, Mach waves produced within the latter part of a single expansion fan are directed at an angle farther to the sideline than those produced at an upstream position within the same fan, leading to converging directions of travel and focusing. A simple model is developed and used to demonstrate the explanatory power of this mechanism with input from a prior computational study.

9:50

4aNS4. Large-eddy simulations of supersonic rectangular jets including screech. Gary Wu (Dept. of Aeronautics and Astronautics, Stanford Univ., Durand Bldg., Rm. 204, 496 Lomita Mall, Stanford, CA 94305, gaojunwu@stanford.edu), Sanjiva K. Lele (Dept. of Aeronautics and Astronautics, Stanford Univ., Stanford, CA), and Jinah Jeun (Ctr. for Turbulence Res., Stanford Univ., Stanford, CA)

Large-eddy simulations (LES) are used to study supersonic jet screech, an aeroacoustic resonance phenomenon found in non-ideally expanded jets. This work specifically analyses screech generation for an under-expanded cold jet exhausted from a converging-diverging rectangular nozzle. With an aspect ratio of 4:1, the nozzle has a design Mach number of 1.44 and was previously tested at Florida State University. The simulations are performed with the compressible flow solver, "CharLES," developed by Cascade Technologies. The meshes are generated based on the computation of Voronoi

diagrams and contain around 40 to 120 $\times$ 10^6cells, targeting a few set points close to maximum screech. The far-field acoustics are obtained using a permeable Ffowcs Williams-Hawkings formulation in the frequency domain. In the noise spectra, screech tones are detected at various observer locations. At the specific tonal frequency, a spatially modulating standing wave pattern is observed in the near-field flow data, which possibly results from the partial interference between oppositely moving hydrodynamic and acoustic waves. Using the LES data, a modal form for the partial standing wave pattern can be obtained. Furthermore, a more in-depth stability analysis can provide insights on the origins and effects of these coherent features associated with screech.

10:05

4aNS5. Analysis of turbulence in convergent-divergent nozzles with fluid insert noise reduction technology. Chitrarth Prasad (Aerosp. Eng., Penn State Univ., University Park, PA) and Philip Morris (Aerosp. Eng., Penn State Univ., 233C Hammond Bldg., University Park, PA 16802, pjm@psu.edu)

The noise generated by tactical fighter aircraft engines is harmful to personnel working in the vicinity of the aircraft and can cause annoyance to communities in the vicinity of air bases. The primary noise sources in supersonic jets are related to the supersonic convection of large-scale turbulent structures, which generate intense noise in the jet downstream direction, and the interaction of the turbulence with the jet's shock cell structure that results in broadband shock-associated noise, which radiates predominantly in the sideline and upstream directions. The use of fluid inserts in the divergent section of variable area exhaust nozzles has been shown experimentally, at small and moderate scale, to reduce noise radiation from both noise sources. To understand the fluid insert noise reduction mechanisms, a Large Eddy Simulation database is developed and analyzed. The focus is on differences between turbulence properties in a baseline and a fluid insert nozzle. The flow database is analyzed using Spectral Proper Orthogonal Decomposition and Doak's Momentum Potential Theory. The latter theory permits the decomposition of the flow into hydrodynamic, thermal and acoustic components. The changes in the acoustic component are related to the observed changes in the radiated noise.

10:20–10:35 Break

Invited Paper

10:35

4aNS6. Summary of acoustic design for H3 launch complex. Wataru Sarae (JAXA, 2-1-1 Sengen, Ibaraki, Tsukuba 305-8505, Japan, sarae.wataru@jaxa.jp), Keita Terashima (JAXA, Tsukuba, Japan), Seiji Tsutsumi (JAXA, Sagamihara, Kanagawa, Japan), Masao Takegoshi (JAXA, Kakuda, Japan), Hiroaki Kobayashi (JAXA, Kanagawa, Japan), and Ayano Mori (JAXA, Kagoshima, Japan)

H3-scaled Acoustic Reduction Experiments (HARE) with a scale of 2.5 % was conducted for the prediction and reduction of lift-off acoustics for the H3 launch vehicle currently under development in Japan. The effect of vehicle elevation and shape of a movable launch platform was studied in the first testing campaign. Then, acoustic reduction based on acoustic shielding and water injection system was also studied in the second testing campaign. In this presentation, results of those testing campaign will be summarized. Then, in addition to the numerical study, acoustic design of the launch complex conducted in the H3 development will be presented.

Contributed Papers

10:55

4aNS7. Decomposition of the Navier-Stokes equations for noise source quantification within turbulent jets. Steven A. Miller (Mech. and Aerosp. Eng., Univ. of Florida, PO BOX 116250, 939 Sweetwater Dr., Gainesville, FL 32611, saem@ufl.edu), Trushant K. Patel, and Weiqi Shen (Mech. and Aerosp. Eng., Univ. of Florida, Gainesville, FL)

Jet noise remains a community annoyance and a major source of hearing damage for military personnel. The physical and mathematical understanding of the noise source are critical for the purpose of its reduction. We decompose the flow-field into a base flow, large-scale spatially coherent structures, small-scale relatively incoherent structures, and associated radiated noise components. The large-scale and fine-scale structures are ascertained via an additional numerical decomposition approach. An acoustic analogy based model is used to predict the noise from the relatively small-scale incoherent turbulence and the large-scale coherent turbulence interacting with the shock-cell structure. We examine these source models combined with the decomposition of large-eddy simulation (LES). We show validated noise predictions of our LES solver with both the Ffowcs Williams and Hawkings equation and our newly developed jet noise model. We evaluate the source models obtained from the acoustic analogy, that are the two-point cross-correlations of the Navier-Stokes equations, to quantify all noise sources. The major advantage of the developed approach is the ease of quantifying both the shock-associated noise and the fine-scale mixing noise sources. Finally, we discuss how our model can be used with existing LES of turbulent flows for noise reduction.

11:10

4aNS8. Experimental study of over-expanded jets with centered thermal non-uniformity. Kyle A. Daniel (Aerosp. and Ocean Eng., Virginia Tech, 460 Old Turner St., Blacksburg, VA 24060, kyled1@vt.edu), David Mayo, Kevin T. Lowe, and Wing Ng (Aerosp. and Ocean Eng., Virginia Tech, Blacksburg, VA)

Recently, there has been increased concern in reducing supersonic jet noise, due in part to stricter community noise standards, renewed interest in supersonic transport over land, and the hearing damaged sustained by Navy personnel operating on the decks of aircraft carriers. In support of this, we propose an examination of the flow and acoustic fields of a supersonic jet with nozzle boundary conditions tailored to produce thermal non-uniformity. Specifically, we aim to study a heated jet with a centered thermal non-uniformity operating at over-expanded conditions. It will be shown through the course of this study that thermal non-uniformity reduces the overall sound pressure level (OASPL) by a maximum of 2.5 dB near the peak noise direction. We hypothesize the reduction in the radiated sound field is related to perturbations which persist beyond the near nozzle region and disrupt the organized structure of acoustically efficient turbulence over large axial distances. This hypothesis is supported by experimental evidence showing a reduced peak Reynolds shear stress, a decoherence in the hydrodynamic field, and a decorrelation of Mach wave structures. These effects are captured using PIV and high-speed schlieren imaging of the density near-field.

Session 4aPA

Physical Acoustics, Engineering Acoustics, Structural Acoustics and Vibration, and Underwater Acoustics: Aqueous Acoustic Metamaterials I

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

Shane W. Lani, Cochair
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Jason J. Smoker, Cochair
NSWCCD, 9500 MacArthur Blvd, West Bethesda, Maryland 20817

Chair's Introduction—9:00

Invited Papers

9:05

4aPA1. Soft acoustic metamaterials: From broadband tunable metagels to directional emission. Nicholas X. Fang (MechE, MIT, 77 Massachusetts Ave., Cambridge, MA 02139, nicfang@mit.edu), Chu Ma, and Xinhao Li (MechE, MIT, Cambridge, MA)

In this invited talk, we report the experimental investigation of hybrid meta-gel, a class of designed hydrogel composites with tunable acoustic properties over broadband frequencies. The meta-gel consists of patterned channels in a tough hydrogel matrix, where air, water, or liquid metal can be purged through the channels to tune the meta-gel's acoustic transmission over broadband frequencies on demand. We show that the acoustic properties of the meta-gel can be tuned by varying the volume ratio of the channels, properties and ratio of the different filler materials with combined experiments, theory, and simulations. We validated that the hybrid metamaterial significantly increased directivity and main lobe energy over a broad bandwidth both numerically and experimentally. The meta-gel enables novel functions such as adjustable imaging regions of ultrasound, demonstrating tangible applications in underwater acoustics and medical imaging.

9:25

4aPA2. Phononic crystal based tunable and nonreciprocal ultrasonic metamaterials. Arup Neogi (Dept. of Phys., Univ. of North Texas, Denton, TX 76203, arup@unt.edu), Hyeonu Heo (Phys., Univ. of North Texas, Denton, TX), Jaehyung Ju (Joint Inst. of Michigan Univ., Shanghai Jiaotong Univ., Shanghai, China), and Arkadii Krokhin (Phys., Univ. of North Texas, Denton, TX)

Phononic crystals with asymmetric scatterers is an ideal platform to control the transmission of acoustic waves. These asymmetric scatterers has been used for nonreciprocal ultrasonic wave propagation in a fluidic medium due to the differential dissipation induced by the scatterers. The non-reciprocity of the wave-propagation depends significantly on the surface properties and the orientation of these individual scatterers. These quasi-periodic phononic structures with asymmetric scatterers can be used to modulate the propagation of the acoustic waves for an active acoustic insulator to a conductive behavior. The dispersion of the ultrasonic wave through the phononic crystal can be used to focus, beam-steer or filter the acoustic wave propagation as well as realize acoustical cavity for multifunctional applications. The localization of sound wave due to these orientation of the asymmetric scatterers has also been observed experimentally.

9:45

4aPA3. Ultra-compact ultrasonic metalens for underwater focusing. Jian Chen (Nanyang Technol. Univ., Singapore, Singapore) and Zheng Fan (Nanyang Technol. Univ., 50 Nanyang Ave., North Spine N3-02c-92, Singapore, Singapore 639798, Singapore, zfan@ntu.edu.sg)

There are significant challenges to design acoustic metasurface in the ultrasonic regime where the wavelength is small. Moreover, the routine design rule based on effective medium approximation is invalid for underwater ultrasound. Our recent development of metasurface provides a great opportunity to solve above-mentioned challenges and the results demonstrate the great potential to apply this metasurface in the field of ultrasonic focusing. Here, we presented a focusing metasurface lens with extreme simplicity (slot structure) and ultra-compact size (deep-subwavelength spacing and thickness). Instead of tuning the acoustic path length individually, we exploited the strong wave couplings between the deep-subwavelength-spaced units for the phase modulation. A microscopic coupled-wave theory was used to predict the phase profile, based on which the metasurface lens for ultrasonic focusing was optimized. Particularly, broadband

focusing was also exhibited due to the non-resonant arrangement of the proposed metasurface lens. The numerical and experimental results agreed well with each other, effectively validating the feasibility of the proposed metasurface lens. The proposed approach provides a feasible avenue for the design of simple and ultra-compact ultrasonic lenses that would be suitable in various fields of ultrasonics.

10:05–10:25 Break

10:25

4aPA4. Wavefront shaping in underwater with soft gradient-index metasurfaces at ultrasonic frequencies. Yabin Jin (I2M, Université de Bordeaux, Bordeaux, France), Raj Kumar (CRPP, Université de Bordeaux, Bordeaux, France), Olivier Poncelet (I2M, Université de Bordeaux, TALENCE, France), Olivier Mondain-Monval (CRPP, Université de Bordeaux, Pessac, France), and Thomas Brunet (I2M, Université de Bordeaux, 351, cours de la libération, Bâitment A4 - I2M/APY, Talence 33405, France, thomas.brunet@u-bordeaux.fr)

Metasurfaces are planar metamaterials with a subwavelength thickness that allows wavefront shaping by introducing in-plane variations, namely, gradients, in the spatial wave response of these flat structures. Most of the recently reported acoustic metasurfaces were designed to control air-borne waves. Here, we report a new class of acoustic gradient-index (GRIN) metasurfaces engineered from soft graded-porous silicone rubber with a high and tunable acoustic index for broadband ultrasonic 3-D wavefront shaping in aqueous media. The functionalities of these soft flat lenses are illustrated through various experiments, which demonstrate beam steering and beam focusing, as well as vortex beam generation in free water. These new GRIN metasurfaces may have important applications in various domains using designed ultrasonic fields (biomedical imaging, industrial non-destructive testing, contactless particle manipulation), since their fabrication is very straightforward with common polymer science engineering. Y. Jin, R. Kumar, O. Poncelet, O. Mondain-Monval, and T. Brunet, *Nat. Commun.* **10**, 143 (2019).

10:45

4aPA5. Aqueous acoustic metasurface for the anomalous reflection of sound. Caleb F. Sieck (Code 7160, U.S. Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, caleb.sieck@nrl.navy.mil), Theodore P. Martin (Excet, Inc., Springfield, VA), James P. Wissman (Code 7160, NRC Res. Associateship Program, U.S. Naval Res. Lab., Washington, DC), Alec Ikei, Jeffrey S. Rogers, Matthew D. Guild, and Charles Rohde (Code 7160, U.S. Naval Res. Lab., Washington, DC)

Ultrathin structures known as acoustic metasurfaces offer the same ability to achieve extreme properties as acoustic metamaterials, yet have the added potential benefit of being very thin (much less than a wavelength in thickness). In this work, we discuss an aqueous acoustic metasurface that utilizes subwavelength non-resonant embedded cavities to achieve anomalous reflection of incident acoustic waves impinging over a wide range of angles. The metasurface was constructed using air-filled cavities in a low-shear matrix material and designed to shift the angle of the reflected wave by 30 deg. The use of a low sound speed matrix material and non-resonant cavities enabled operation over a wider range of frequencies and incident angles while maintaining subwavelength thickness. The cavity heights were varied to achieve 0 to 2π phase shifts in a unit cell according to the generalized form of Snell's Law. The fabricated structure was mounted on a brass plate and experimentally tested in water. Very good agreement was found between finite element models and the experimental ultrasonic tank tests of the aqueous acoustic metasurface. [Work supported by the Office of Naval Research.]

11:05

4aPA6. Broadband superabsorption of waterborne acoustic waves by bubble metascreens. Maxime Lanoy (Dept. of Phys. and Astronomy, Univ. of MB, Winnipeg, MB R3T 2N2, Canada, maxime.lanoy@espci.fr), Valentin Leroy (Laboratoire Matière et Systèmes Complexes, Université Paris-Diderot, CNRS (UMR 7057), Paris, France), Steven Squire, Anatoliy Strybulevych, Reine-Marie Guillermic, Eric J. Lee (Phys. and Astronomy, Univ. of MB, Winnipeg, MB, Canada), Fabrice Lemoult, Arnaud Tourin (Institut Langevin, ESPCI Paris, CNRS, PSL Univ., Paris, France), and John H. Page (Phys. and Astronomy, Univ. of MB, Winnipeg, MB, Canada)

Absorption of acoustic or mechanical waves is an important challenge for various applications such as noise insulation, stealth coating, seismic event mitigation and ultrasonic testing. In order to absorb sound, one needs to introduce a medium with sufficient dissipation but without significant reflection of the incoming wave. Ideally, the absorber should also be thin and light, a goal that may be realized through the use of a thin 2-D metamaterial, or metalayer. The conventional way of achieving strong absorption in a thin metamaterial is to exploit low-frequency resonant inclusions. However, most resonant structures have an intrinsically narrowband response, making it difficult to attain broadband absorption in a deeply subwavelength-thick meta-layer, and making it necessary to devise larger structures to increase the bandwidth. For waterborne acoustic waves, an exception to this common situation can be achieved through the fabrication of bubble metascreens, which consist of a single layer of bubble inclusions embedded in a soft solid. In this presentation, we re-visit the optimization of such bubble metascreens and show that, despite being resonance-based, near-perfect absorption is possible over a very wide frequency range even when the metalayer is ultrathin.

11:25

4aPA7. Non-Hermitian complimentary acoustic metamaterials for imaging through skull. Steven R. Craig (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta 30318, Georgia, scraig32@gatech.edu), Phoebe J. Welch (Mech. Eng., Georgia Inst. of Technol., Atlanta, Georgia), and Chengzhi Shi (Mech. Eng., Georgia Inst. of Technol., Berkeley, CA)

Complimentary metamaterials are designed to facilitate full acoustic transmission through highly mismatched impedance barriers. Applied to biomedical acoustics, complimentary metamaterials theoretically enable unidirectional acoustic transmission through the skull to realize noninvasive brain imaging and facilitate neural ultrasound therapies. Previously designed complimentary acoustic metamaterials relied on the realization of negative material parameters opposite to that of the skull to cancel the impedance mismatch effect. However, much of the incident acoustic energy is dissipated due to the skull's internal porosity, preventing full energy transmission. To resolve this issue, we propose to counteract the skull's acoustic attenuating effects with a non-Hermitian complimentary metamaterial

(NHCMM) having an active gain circuit to achieve high acoustic transmission at high frequencies. In addition to the full wave simulations showing near perfect transmission from either side of the skull, the NHCMM preserves imaging information for objects inside the skull, and enhances the focused acoustic energy used for focused ultrasound therapies. By facilitating two-way acoustic transmission through the skull with NHCMMs, we lay the foundation for noninvasive brain imaging and treatment for neural disorders.

Contributed Paper

11:45

4aPA8. Experimental realization of aqueous double negative metamaterials in near-megahertz range. Jiaying Wang (Dept. of NanoEng., Univ. of California, San Diego, 9500 Gilman Dr., 0401, La Jolla, CA 92093-0401, jiw372@ucsd.edu), Florian Allein, Nicholas Boechler, James Friend (Dept. of Mech. and Aerosp. Eng., Univ. of California, San Diego, La Jolla, CA), and Oscar Vazquez-Mena (Dept. of Nanoeng., Univ. of California, San Diego, La Jolla, CA)

Ultrasound is strongly attenuated and aberrated by the skull, principally due to the complex structure and acoustic impedance mismatch of skull which generates strong scattering. Analytical models and finite element methods have shown that negative index metamaterials matching skull properties can enable the transmission of US which enhances bidirectional transmission of ultrasonic signal for imaging. However, it is difficult to realize double-negative metamaterials in near-megahertz range experimentally because of the limitation of fabrication and characterization methods. Here we show the experimental realization of waterborne double negative acoustic metamaterials by coupling micrometer-size membrane-based negative density metamaterials with Helmholtz resonator-based negative modulus materials. Numerical simulation and experimental verification consistently exhibit the double negative behavior between 230 kHz and 410 kHz. We demonstrated the feasibility of creating ultrasonic-range metamaterials for non-invasive ultrasound imaging and neuron stimulation.

Session 4aSA

Structural Acoustics and Vibration and Computational Acoustics: Computational Methods for Mid-Frequency Structural Acoustic Problems

Anthony L. Bonomo, Cochair
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David Raudales, Cochair
Naval Research Lab, 1420 W Abingdon Drive Apt 126, Alexandria, Virginia, United States

Chair's Introduction—8:00

Invited Papers

8:05

4aSA1. The pollution effect and novel methods to reduce its influence in the mid-frequency range. Jerry W. Rouse (Analytical Structural Dynam., Sandia National Labs., P.O. Box 5800, Albuquerque, NM 87185, jwrouse@sandia.gov) and Timothy F. Walsh (Simulation and Modeling Sci., Sandia National Labs., Albuquerque, NM)

The classical Galerkin formulation is nearly ubiquitous in production-scale finite element codes. The formulation uses effectively the same space of polynomials for both the solution (trial) and weighting (test) functions. Further, the standard h refinement method of decreasing element size to achieve convergence is equally universal, typically involving only linear and second order polynomial elements. For geometries that are several wavelengths in size the above methodology is sufficient for accurate predictions. However, extending beyond several wavelengths, ever-present interpolation and pollution errors degrade the predictions at a linear and nonlinear rate, respectively. As the number of wavelengths in the geometry increases, the nonlinear (pollution) term eventually dominates the approximation error. Prediction accuracy can be maintained by correspondingly increasing the number of elements per wavelength but at a cost of increasing computational need. This talk shall present an overview of interpolation and pollution errors and their effects, as well as a number of cutting-edge methodologies developed to overcome their influence in the mid-frequency range.

8:25

4aSA2. On using geometry-based high-order finite element approximations for mid-frequency response of fluid-loaded elastic structures. Saikat Dey (NRL, 4555 Overlook Ave. SW, Washington, DC 20375, saikat.dey@nrl.navy.mil)

Accurate solution of the exterior scattering and radiation problems in the mid-frequency regime remains of high interest and a computationally challenging task. For wave-dependent problems with large dispersion errors, high-order approximations provide several advantages, including increased accuracy and error convergence rate compared to low-order methods. The effective use of high-order finite element type approximations require special attention to geometry. In particular, one must use high-order (curved) meshes in order to preserve the accuracy and convergence advantages. We present recent developments in the use of geometry-based high-order approximations for computing scattering response of fluid-loaded elastic structures submerged in an acoustic medium. We present examples with validation against experimental data to demonstrate the effectiveness of these techniques as well as highlight future technical challenges to push to the higher ends of the mid-frequency regime.

Contributed Paper

8:45

4aSA3. Broadband monostatic acoustic scattering simulations of underwater unexploded ordnance using time-domain spectral-elements. Alexis Bottero (Marine Physical Lab, Scripps Inst. of Oceanogr., 9500 Gilman Dr., La Jolla, CA 92093, abottero@ucsd.edu), Earl G. Williams (Naval Res. Lab., Washington, DC, DC), William A. Kuperman (Marine Physical Lab, Scripps Inst. of Oceanogr., La Jolla, CA), and Sandrine Rakotonarivo (Aix-Marseille Univ., Marseille, France)

Acoustic detection of unexploded ordnance (UXO) that contaminate the world's waterways is vital. Critical to the implementation of detection methods is the numerical simulation of the scattering response of these targets using finite element methods. We introduce a parametric approach based on the time-domain spectral finite element method (SEM). This technique allows for the computation of broadband acoustic response of complex heterogeneous 3-D fluid-solid objects, in particular a 5 inch rocket UXO used for this study. The scattered field is obtained at arbitrary distances using the Kirchhoff–Helmholtz integral. The main interest of the SEM is that it is particularly adapted to high-performance computing (CPU or GPU). Contrary to most of the other finite element methods it scales perfectly; using twice as many processors leads to a halving of computing time. In addition, working in the time domain is a direct simulation of common sonars and experiments. The method is first benchmarked against the commercial finite-element code COMSOL on the monostatic response of a rigid target over a full 180 deg. Finally, results obtained for the 5 inch rocket are compared to actual measurements obtained in the NRL underwater acoustics tank facility. [Work partially supported by the Office of Naval Research.]

9:00

4aSA4. Efficient, wide-band coupled structural-acoustic computations combining time and frequency domain finite elements/equivalent sources. John B. Fahnline (ARL / Penn State, P.O. Box 30, State College, PA 16804-0030, jbf103@arl.psu.edu)

Time and frequency domain computations are complementary, and here, the overall goal is to combine them to generate wide-band solutions of coupled structural-acoustic problems. Modal frequency response computations are efficient for low frequencies but become inefficient as the number of basis functions required to represent the response grows large. Modal formulations are required for large problems because the coupled equation system becomes densely populated when acoustic pressure forces are included. In contrast, the equation system remains sparsely populated in time domain formulations when the time step size is chosen appropriately, allowing large problems to be solved efficiently in terms of nodal degrees-of-freedom. However, the analysis times depend linearly on the number of time steps and a large number are generally required to represent the ringing response of low frequency modes. Here, the number of time steps is limited by applying an exponential taper to the time domain solution and truncating the computations. This reduces leakage effects, but does not affect components of response that have damped out before it is applied. Examples are given to illustrate the computations and the tradeoff between number of basis functions in the modal frequency response computations and number of time steps in the transient analyses.

9:20

4aSA5. A substructuring approach based on mechanical admittances to solve vibro-acoustic problems in the mid-frequency range. Valentin Meyer (Naval Group, 199 av. Pierre-Gilles de Gennes, Ollioules 83190, France, valentin.meyer@naval-group.com) and Maxit Laurent (1. Univ Lyon, INSA–Lyon, Laboratoire Vibrations-Acoustique (LVA), Villeurbanne, France)

In the low frequency range, discretization methods such as the Finite Element Method are the preferred techniques to solve vibro-acoustic problems. The main limitation of such approaches is that the mesh needs to be refined when the frequency increases, leading to prohibitive calculation costs. This paper presents the Condensed Transfer Function (CTF) method for substructuring vibro-acoustic problems. The problem is split in several structural, acoustic or vibro-acoustic subsystems that are studied individually before being assembled. The vibro-acoustic behavior at the boundaries of the subsystems is expressed in terms of mechanical admittance for structures and acoustic impedance for fluid domains. A set of condensation functions is used as a basis to express the values at the boundaries. The coupling between subsystems is written through the continuity equations of mechanics. Compared to classical reduction methods such as Craig-Bampton, this approach does not require the prior knowledge of the modal basis of the subsystems. Also, the CTF method allows coupling subsystems that are described by different techniques, i.e., analytically, numerically or experimentally. The principle of the method will be recalled and example of applications will be given. In particular, numerical and experimental results on a stiffened cylindrical shell will be compared.

9:40

4aSA6. Approximate models for the structural-acoustic response of elastic barriers suitable for hybrid analytical-computational techniques. Mauricio Villa (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC) and Donald B. Bliss (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Mech. Eng., 148B Hudson Hall, Durham, NC 27708, donald.bliss@duke.edu)

Exact and approximate models are considered for the fluid-loaded response of finite panels driven by an obliquely incident acoustic wave. To evaluate the acoustic back-coupling of these fluid-loaded structures, the kernel function of the integral-differential equation is approximated using new and efficient trigonometric representations for small arguments. This representation allows for accurate matching to the asymptotic expansions with very few terms. The approach efficiently treats arbitrary boundary conditions by interpreting them as deviations from a simply supported panel corrected by a smooth force distribution. Simple, closed-form approximate models that represent the dominant radiation mechanism are also pursued. The fluid-loaded reverberation of the panel is modeled by approximating the inertial and radiation damping effects of the bounding fluids based on an understanding of the radiation mechanisms for subsonic and supersonic flexural waves. Comparison to benchmark solutions confirms the simple radiation models successfully describe the dominant characteristics of the structural-acoustic response. These models are demonstrated to work extremely well in the mid to high frequency range. Their success has important implications for the creation of hybrid analytical/computational models that are efficient and accurate for multi-dimensional mid-frequency structural acoustics.

10:00

4aSA7. Scaled boundary finite element method for mid-frequency interior acoustics. Sundararajan Natarajan (Dept. of Mech. Eng., Indian Inst. of Technol. Madras, Rm. 207, Machine Design Section, Chennai, Tamil Nadu 600036, India, snatarajan@iitm.ac.in) and Chandramouli Padmanabhan (Mech. Eng., Indian Inst. of Technol. Madras, Chennai, Tamil Nadu, India)

In this talk, a semi-analytical framework, based on the scaled boundary finite element method (SBFEM), is proposed, to study interior acoustic problems in the mid-frequency range. The SBFEM shares the advantages of both the finite element method (FEM) and the boundary element method (BEM). Like the FEM, it does not require the fundamental solution (Green's function) and similar to the BEM only the boundary is discretized, thus reducing the spatial dimensionality by one. The solution within the domain is represented analytically, while on the boundary, it is represented by finite elements. Different choices of boundary representations, such as Lagrange and NURBS description will be discussed. The proposed framework is validated using closed-form solutions and direct comparisons are made with conventional FEM based on Lagrangian description; this will be demonstrated using two two-dimensional cavities available from the literature. The improved accuracy and reduced computational time can be attributed to the semi-analytical formulation combined with the boundary discretization.

10:20–10:35 Break

10:35

4aSA8. Comparison between finite element and hybrid finite element results to test data for the vibration of a production car body. Nickolas Vlahopoulos (Univ. of Michigan, 2600 Draper Dr., Ann Arbor, MI 48109, nickvl@umich.edu) and David Sander (Virtual Vehicle, Graz, Austria)

The Hybrid Finite Element Analysis (Hybrid FEA) method is based on combining conventional Finite Element Analysis (FEA) with analytical solutions and energy methods for mid-frequency computations. The method is appropriate for computing the vibration of structures which comprises stiff load bearing components and flexible panels attached to them and for considering structure-borne loadings with the excitations applied on the load bearing members. In such situations, the difficulty in using conventional FEA at higher frequencies originates from requiring a very large number of elements in order to capture the flexible wavelength of the panel members which are present in a structure. In this presentation a three-way comparison will be offered for the vibration of a production vehicle body in the frequency range 200 Hz–1000 Hz. Six different excitation locations are utilized (one at a time); for each excitation the mobility of five reference body points on the load bearing members, and the mobility of eight flexible vehicle panels is measured and the measurements are compared with both conventional FEA results and with Hybrid FEA computations. Discussion about the development of the Hybrid FEA model and the correlation of both numerical solutions to the test data will be presented.

10:55

4aSA9. Dynamical energy analysis—An efficient mesh based simulation tool for mid-to-high frequency structure-borne sound calculations. Gregor Tanner (School of Mathematical Sci., Univ. of Nottingham, University Park, Nottingham NG7 2RD, United Kingdom, gregor.tanner@nottingham.ac.uk)

Modelling the vibro-acoustic response of mechanical built-up structures is a challenging task—especially in the mid to high frequency regime—even with the computing powers available today. Standard modelling tools such as Finite Element Methods (FEM) are not scalable to high frequencies due to the prohibitive increase in model size. Typical high frequency methods such as the Statistical Energy Analysis (SEA) on the other hand, often lack details of the structure and require care in the setting up the model. In this talk, I will review the Dynamical Energy Analysis (DEA)—a method bridging the gap between the low and very high frequency range. DEA describes the energy flow in a complex structure keeping the directional information of energy propagation. It can be set up directly on meshed structures thus providing detailed spatial information about the vibrational energy of a built-up structure of high complexity. The response of coupling coefficients between sub-components are obtained through local wave models integrated into the global DEA treatment. The computational cost of DEA is frequency independent making it possible to get results from the mid-to-high frequency regime. The method can be extended to include sound radiation calculations in a post-processing step. The new method has been validated both using FEM computations and experiments in collaboration with Yanmar Co., Ltd., a Japanese tractor manufacturer. Further applications in the car, aerospace and ship building industry will be presented.

Contributed Papers

11:15

4aSA10. Dynamical energy analysis applied to real-world structures—New developments. Martin Richter (School of Mathematical Sci., Univ. of Nottingham, University Park, Nottingham NG7 2RD, United Kingdom, martin.richter@nottingham.ac.uk), Gregor Tanner (School of Mathematical Sci., Univ. of Nottingham, Nottingham, United Kingdom), and David J. Chappell (School of Sci. and Technol., Nottingham Trent Univ., Nottingham, United Kingdom)

Dynamical Energy Analysis (*DEA*) has been established as a mesh-based ray tracing method working at mid-to-high excitation frequencies for structure-borne sound on complex vehicle structures. Unless standard ray-tracing used, for example, in room acoustics and following long rays through a series of specular reflections, DEA is based on an integral representation by (a) reformulating the energy transport problem in terms of ray densities and linear matrix equations and by (b) applying the ray tracing directly on preexisting meshes used also in, for example, FEM calculations. The advantage is that DEA can be used as a drop-in replacement for FEM tool working on the same meshes but determining results at very high frequencies. In order to achieve this, a so-called transfer matrix has to be created and later solved for the energy equilibrium distribution. In this presentation, we address recent developments in making DEA an efficient black-box tool for mid- to high-frequency vibrations including convergence issues and suitable preprocessor applications, sound radiation from DEA calculations as well as verification and validation studies.

11:30

4aSA11. An efficient model order reduction of frequency-dependent double panel systems. Thiago Cavalheiro (Dept. of Mech. Eng., Universidade Federal de Santa Catarina, Laboratório de Vibrações e Acústica/UFSC, Florianópolis, Santa Catarina 88040900, Brazil, thiago.cavalheiro@lva.ufsc.br), Marcos S. Lenzi (Dept. of Civil Eng., Universidade Federal de Santa Catarina, Florianópolis, Santa Catarina, Brazil), Gustavo C. Martins (Flow Energia, Florianópolis, Santa Catarina, Brazil), and Arcanjo Lenzi (Dept. of Mech. Eng., Universidade Federal de Santa Catarina, Florianópolis, SC, Brazil)

This paper treats a typical industrial vibroacoustic problem, consisting of a double panel under forced vibration which radiates to the far-field. A finite element model is proposed, where both structural links and porous material are represented as equivalent models. The addition of frequency-dependent elements, as mechanical links and porous material in the cavity, leads to a balance between airborne and structural-borne transmission through panels in the mid-frequency range, which demands a suitable model to represent these phenomena. In order to cope with such large-scale models, a dedicated parametric model order reduction is proposed. The frequency-dependent terms are separated from the system matrices, allowing an offline reduction of the system. The Second-Order Arnoldi procedure is employed to calculate the reduced bases at multiple frequencies, chosen by a rational interpolation scheme. The expansion frequencies are selected by a novel adaptive windowing algorithm, which aims at minimizing the number of frequencies within the frequency range. The proposed approach has shown to be effective in reducing significantly the computational time while maintaining precision compared to the direct method.

11:45

4aSA12. Further exploration of model truncation to extend the applicability of the finite element method to higher frequencies. Anthony L. Bonomo (Naval Surface Warfare Ctr., Carderock Div., 9500 MacArthur Blvd, West Bethesda, MD 20817, anthony.l.bonomo@navy.mil)

In a previous presentation, the use of model truncation to extend the applicability of the finite element method to higher frequencies was explored. This talk furthers that exploration and focuses on the development of an easy-to-implement alternative to the perfectly matched layer for the truncation of semi-infinite problems with coupled structural and acoustic domains. Numerical results showing the efficacy of the proposed method are discussed and a method to bound the error that results from treating a large finite structural as semi-infinite is considered. It is hoped that the proposed method helps facilitate extension of the finite element method to mid-frequency applications. [Work supported by ONR.]

THURSDAY MORNING, 5 DECEMBER 2019 CROWN, 8:00 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: Speech Production (Poster Session)

Marc Garellek, Chair
Dept. of Linguistics, UCLA, Los Angeles, California 92093

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors in this session an opportunity to see other posters, contributors 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 to 12:00 noon.

Contributed Papers

4aSC1. Acoustic manifestations of "narrow focus" in Apurímac Quechua vowels. Faisal Alkahtani (Univ. of Delaware, 716 Village Cir, Apt. C, Newark, DE 19713, kahtani@udel.edu)

This paper investigates the acoustic manifestations of "narrow focus" on the vowels of Apurímac Quechua. Specifically, the three acoustic correlates of duration, intensity, and mean F0 in the vowels of stressed syllables were measured to see how narrow focusis expressed and if it would have a significant enhancing effect on the target vowels /a/, /i/, and /u/. 180 tokens were used in this experimental study, 90 in "focus" condition and 90 in "non-focus" condition (30 for each vowel in both conditions). Paired t-tests were conducted for each correlate, and the results show that duration is significantly longer ($p < 0.01$) in focus condition (mean = 47.22 ms, SD = 10.84) compared to non-focus condition (mean = 47.22 ms, SD = 10.84). The same pattern is found for intensity where vowels in focus condition (mean = 66.44 dB, SD = 4.85) were a significantly louder ($p < 0.01$) than the vowel in non-focus condition (mean = 63.02, SD = 5.27). Finally, the mean F0 for vowels in focus condition (mean = 135.41 Hz, SD = 16.45) was significantly higher ($p < 0.01$) than the non-focus condition (mean = 117.41). While all correlates show significant increase as a result of being in a focus condition, the duration correlate did not cross the "Just Noticeable Difference" (JND) threshold of 10 ms. On the other hand, intensity and F0 crossed the JND of 3 dB and 1 Hz, respectively. These findings suggest that Apurímac Quechua is a syllable-timed language that does not rely on duration as a primary acoustic correlate of "narrow focus."

4aSC2. Using H1 instead of H1–H2 as an acoustic correlate of glottal constriction. Yuan Chai (Dept. of Linguist, Univ. of California San Diego, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, yuc521@ucsd.edu) and Marc Garellek (Linguist, Univ. of California San Diego, La Jolla, CA)

Bickley (1982) proposed H1–H2 as an index of the stronger H1 observed for breathy vowels compared to modal ones. Because harmonic amplitudes also vary by overall sound pressure level (SPL), H1 amplitude was normalized to that of H2. Studies have also shown that H1–H2 is well perceived and correlates with glottal open quotient, and thus can be used to infer the articulations leading to changes in perceived quality. Yet in some cases, H1–H2 does not distinguish two voice qualities at statistically significant levels. We claim that this might be due not to the same open quotient between two voice qualities, but to the fact that H1–H2 can be especially noisy to estimate: for example, when f0 is irregular (as in creaky voice qualities), H1–H2 often has large variance that isn't well modeled statistically. Using several corpora, we show that H1 alone (while controlling for SPL) allows for stronger statistical differentiation of voice categories than H1–H2. We further confirm that H1 can stand in place of H1–H2 as a correlate of glottal constriction. When differentiating voice qualities acoustically, researchers should consider measuring H1 (while controlling for differences in overall SPL) instead of, or in addition to, H1–H2.

4aSC3. The neutralization of Xiapu Min tones in disyllabic tone sandhi forms. Yuan Chai (Dept. of Linguist, Univ. of California San Diego, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, yuc521@ucsd.edu)

Tone sandhi can result in both incomplete and complete tone neutralization. Chien and Jongman (2018) proposed that unproductive (i.e., lexicalized) tone sandhi (as in Taiwan Min) tends to have complete neutralization; productive tone sandhi usually shows incomplete neutralization (as in Mandarin; Zhang and Lai, 2010). The current study tests whether tone sandhi is phonetically neutralized in Xiapu Min, an Eastern Min variety from Fujian, China (Wen, 2015). Xiapu Min has seven tones, five of which are smooth tones (44, 11, 42, 35, 23) and two of which are checked tones (5, 2). Tone sandhi occurs in disyllabic words and results in neutralization (e.g., Tone 42, 35, and 5 are neutralized to Tone 55 when word-initial.) We test the completeness of this neutralization by investigating potential differences in F0, duration, and phonation, which is measured acoustically using H1*–H2* and HNR, and articulatorily using electroglottographic open quotient.

We also determine the productivity of Xiapu Min by testing whether tone sandhi emerges in nonce words. The results will illustrate whether the tonal neutralization is complete in Xiapu Min real words and will bring further data to bear on the relation between sandhi productivity and the completeness of neutralization.

4aSC4. The influence of speech prosodic structure and syllable composition on consonant VOT in Japanese. Xi S. Chen (Chinese and Bilingual Studies, Hong Kong Polytechnic Univ., Hung Hom, Kowloon, Hong Kong 00000, Hong Kong, skyexi.chen@connect.polyu.hk), Si Chen, and Bei LI (Chinese and Bilingual Studies, Hong Kong Polytechnic Univ., Hong Kong, Hong Kong)

This research elaborates the joint influences of the position relative to the accented prosodic component and syllable composition on the stops VOT in Japanese. Monosyllables and disyllables start with three types of plosives, namely bilabial, alveolar, and velar, are studied on different positions in prosodic sentences—pre-focus, on-focus, and post-focus. Results show that narrow and contrastive focus prolong the voiced on-focus stops (including bilabial, alveolar, and velar) which bear positive values in broad-focus sentences; but compress the on-focus voiceless consonants (including bilabial, alveolar, and velar) which have negative values neutrally. On the position of pre-focus, voiced consonants shorten the VOT towards to zero, especially for the velar voiced consonants; the voiceless ones follow this pattern exactly. On the position of post-focus, noticeably, the voiced and the voiceless plosives always show the opposite patterns—both of narrow and contrastive focus elicit a longer VOT on following voiced stops but reduce the VOT of voiceless consonants following them. The voiced alveolar show the most apparent compression at pre-focus, which always close to zero. Kinds of results describe significant interaction influences among different factors that bear the potential to change consonant VOT in Japanese.

4aSC5. Analysis of breathiness as an acoustic correlate of nasality in Hindi. Pamir Gogoi (Dept. of Linguist, Univ. of Florida, Gainesville, FL 32611, pgogoi@ufl.edu), Ayushi Pandey (Dept. of Cognit. Sci., Johns Hopkins Univ., Baltimore, MD), and Ratree Wayland (Linguist, Univ. of Florida, Gainesville, FL)

This project investigates the perceived breathiness of nasal segments than non-nasal sonorants like [l] in Hindi. Shared acoustic correlates between nasality and breathiness (like wider F1 bandwidth, appearance of additional poles and zeros in the acoustic spectrum) lead to perceptual consequences that both allow erroneous perception of breathiness as nasality (Ohala and Busà, 1995) or enhanced perception of nasality (Arai, 2006). Breathiness has been observed to co-occur with nasality in French (Styler, 2017) and some Yi languages (Garellek *et al*., 2016). Effects of breathiness on perception of nasality have also been reported in Hindi (Ohala and Ohala, 1993)—vowels that were produced with a relatively open glottis were perceived to be more nasal than their non-breathy counterparts. Based on these findings, we hypothesized that Hindi speakers use breathiness for enhanced perception of nasality. This was tested using the following acoustic parameters of breathiness: Difference between 1st and 2nd harmonic (H1-H2), Cepstral Peak Prominence (CPP), Spectral tilt and Harmonic to Noise Ratio (HNR). The results demonstrated that the nasals were significantly breathier than the non-nasal sonorants, suggesting that breathiness might be used as a cue to enhance the perception of nasality in Hindi.

4aSC6. San Diego accents: A preliminary investigation into the vowel spaces of young males and females. Erica Gold (Dept. of Linguist and Modern Lang., Univ. of Huddersfield, Queensgate, Huddersfield HD1 3DH, United Kingdom, e.gold@hud.ac.uk) and Kate Earnshaw (Linguist and Modern Lang., Univ. of Huddersfield, Huddersfield, United Kingdom)

This study investigates the vowel spaces of eight native San Diego speakers (four males and four females) that are: born and raised in San Diego County, only lived outside San Diego for a maximum of four years, English dominant speakers, Caucasian, and aged between 23 and 32 (mean = 28.6 years old). All participants were recorded using handheld Zoom H2n audio recorders. Participants read a word list of 74 monosyllabic words, 80 sentences (mean = 6.5 words per sentence), and participated in a short 20-minute sociolinguistic interview about themselves and the San Diego area. For the purposes of this study, F1–F3 were measured in Praat for all vowels in the read speech of the eight participants using dynamic formant measurements at 10% intervals across each vowel. Results are normalised across all speakers, and then analyzed in Excel and R. Although this is a preliminary study, this investigation provides some of the first acoustic-phonetic results looking at the San Diego accent in young males and females.

4aSC7. Prominence and tone in the Koasati nominal system. Matthew K. Gordon (Dept. of Linguist, UC Santa Barbara, Santa Barbara, CA 93106, mgordon@linguistics.ucsb.edu) and Jack Martin (College of William & Mary, Williamsburg, VA)

This paper presents results of an acoustic study of tone and prominence in nouns in Koasati, an endangered Muskogean language spoken in Louisiana and Texas. Certain nouns in Koasati are lexically marked for tone whereas, in most nouns, the first syllable is impressionistically most prominent (Gordon *et al*. 2015). In the present study, a list of Koasati nouns, comprising both words with and words without lexical tone, was recorded from six speakers in order to explore the phonetic realization of lexical tone and default word-level prominence. Target words, which ranged from two to six syllables in length, were embedded in a carrier phrase and repeated twice by each speaker. Duration, intensity, and several F0 measurements (mean, maximum, time of maximum, minimum, time of minimum) were logged for segments in the target words. Results indicate that the lexical tone is a LH bitonal sequence, which phonetically varies in its timing and scaling as a function of multiple factors including syllable type, following consonant, and speaker. The default prominence on the first syllable is associated with increased intensity and higher F0 (but not greater duration) and may be interpreted as either word-level stress or a phrase-level intonation effect. [Work supported by NSF.]

4aSC8. A simplification analysis of Andalusian aspirated stop clusters. Duna Gylfadottir (Linguist, Univ. of BC, Totem Field Studios 228, 2613 West Mall, Vancouver, BC V6T 1Z2, Canada, gb.gylfa@gmail.com)

Weakening or "aspiration" of syllable-final /s/ is a widespread phenomenon in Spanish. When /s/ precedes a voiceless stop in varieties with /s/-aspiration, the result is normally a pre-aspirated stop [ht]. Some speakers in southern Spain now realize /s/ + stop (sC) clusters as post-aspirated stops, particularly in Western Andalusian Spanish (WAS) [Ruch and Harrington, *J. Phonics* **45**(1), 12 (2014)]. Previous studies have analyzed this change as a timing realignment [Parrell, *J. Phonics* **40**(1), 37 (2012)] or articulatory overlap [Torreira, *JIPA* **42**(1), 49 (2012)]. The current study is the first to investigate sC clusters in naturalistic data. The data consist of 34 sociolinguistic interviews with WAS speakers. Preliminary results based on 10 speakers indicate a negative correlation between closure duration and VOT, and lower overall duration of post-aspirated clusters. Affrication of /ht/ to [t^s] occurred at a rate of 7%, and deletion of the vowel in 4% of /ht/ tokens in vowel-initial words (e.g., [t^ha] for está), a phenomenon not reported in laboratory speech. We argue that low-level gestural realignment cannot account for the naturalistic variability in sC clusters, and argue for an account involving simplification to a single post-aspirated stop.

4aSC9. Disyllabic word context effects on cue weighting in Mandarin tone perception and production. Xiaojuan Zhang (Xi'an Jiaotong Univ., 28 Xianning Rd (W), Xi'an, Shaanxi 710049, China, xiaojuan.110577@stu.xjtu.edu.cn) and Bing Cheng (Xi'an Jiaotong Univ., Xi'an, Shaanxi, China)

This study aims to examine effects of the disyllabic tonal context on native Chinese speakers' cue weighting for perceiving and producing target Mandarin Tone 2 and Tone 3 that share similar pitch contours. The speech materials were disyllabic Mandarin words, matched by word frequency and familiarity. Acoustic manipulations for the perception test included four key parameters, including fundamental frequency (F0) at tone onset, F0 at turning point, the timing of turning point and F0 at tone offset, which characterized the pitch contours of Tone 2 and Tone 3. Thirty native Chinese speakers participated in this study. These synthetic disyllabic stimuli were

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used in an identification task. Production of the target tones in the disyllabic context was also recorded from each participant. Cue weights for the key acoustic parameters were assessed with logistic regression analyses and linear discriminant analyses to model the extent to which each acoustic cue predicted category membership in the perception and production data. The results demonstrated strong perceptual weighting sensitivity to the disyllabic context in native Chinese speakers. The results are discussed with respect to the dynamics of tone perception and production in word contexts and potential implications for Chinese as a second language learners.

4aSC10. Acoustic and aerodynamic variation in voiced plosives. Wendy Herd (MS State Univ., 2004 Lee Hall, Drawer E, MS State, MS 39762, wherd@english.msstate.edu)

This study investigates variation in the production of word-initial /b, d, g/ by native speakers of English in Mississippi. Forty speakers were recorded reading a short passage and a series of words in isolation. As previously found in this geographic region, phonetic realizations of /b, d, g/ included both short positive and long negative VOTs, with speakers who identified as African American producing plosives with negative VOTs more often than those who identified as Caucasian American; however, negative VOTs were observed among both groups of speakers. Amplitude trajectories and spectral characteristics during the voice bar of the plosives with negative VOTs suggest that phonetic realizations of /b, d, g/ included both prevoiced and prenasalized plosives. As with frequency of negative VOTs, prenasalization was observed among both groups but was higher among African American participants than Caucasian American participants. Nasal airflow data collected from a subset of the participants confirmed that plosives with nasal-like spectral characteristics were produced with greater nasal airflow than those lacking those characteristics. Converging acoustic and aerodynamic evidence indicate that /b, d, g/ variants in Mississippi include unaspirated plosives, prevoiced plosives, and prenasalized plosives. These results also provide evidence of socio-ethnic variation in the production of /b, d, g/.

4aSC11. Centering diphthongization in Hui Chinese dialects. Fang Hu (Inst. of Linguist, Chinese Acad. of Social Sci., 5 Jian Guo Men Nei St., Beijing 100732, China, hufang@cass.org.cn) and Minghui Zhang (East China Normal Univ., Shanghai, China)

This paper reports centering diphthongization in three Hui Chinese dialects. It argues that centering diphthongization serves as clear evidence that monophthongs and diphthongs are not a dichotomy, but a continuum, and thus supports a dynamic theory of vowel production (Hu, 2017). Centering diphthongs in Hui dialects occur as an intermediate vowel category between monophthongs and diphthongs. Fine-grained phonetic details reveal that the process of centering diphthongization is gradient in the three Hui dialects. The Yi county dialect is at an early stage: centering diphthongization is essentially a phonetic process of adding an additional neutralized [ɐ]-like production to the distinctive vowel element, namely [i u y e o] > [iːɐ uːɐ yːɐ eːɐ oːɐ], respectively. The Qimen dialect is at a phonologically developed stage: monophthongs and diphthongized vowels contrast in dynamics, namely the static [i u y] versus the dynamic [iːɐ uːɐ yːɐ], but diphthongized vowels have a temporal structure different to that of plain diphthongs. The Xiuning dialect is at a final stage: monophthongs and diphthongized vowels contrast in dynamics, and diphthongized vowels exhibit a temporal organization similar to that of plain diphthongs.

4aSC12. Voice quality of coarticulated Mandarin tones. Yaqian Huang (Dept. of Linguist, Univ. of California San Diego, 9500 Gilman Dr. #0108, La Jolla, CA 92093-0108, yah101@ucsd.edu) and Yuan Chai (Univ. of California San Diego, La Jolla, CA)

Tonal coarticulation often induces changes in F0 (Xu, 1997; Brunelle, 2009) as well as voice quality (DiCanio, 2012). In Mandarin, voice quality covaries with F0, such that when the F0 is higher or lower than the mid-range pitch, the voice quality often becomes creakier (Kuang, 2017). However, fewer studies have examined how coarticulation can affect a tone's voice quality. Using audio and electroglottography (EGG), this study investigates how F0 and voice quality of Mandarin tones are affected by coarticulation. The stimuli consisted of trisyllabic Mandarin compounds, where each of the four Mandarin tones is flanked by varying tones, for a full range of contextual variation. We expect that the F0 of a high tone (e.g., Tone 1) would become lower when flanked by a low tone (e.g., Tone 3), whereas the F0 of a low tone would become higher when adjacent to a high tone. Moreover, the voice quality of Tone 3 that is raised in F0 should become more modal, if voice quality depends on F0. Discussion of results will emphasize how F0 and voice quality of Mandarin tones can vary according to tonal context and how articulation (as measured by EGG) informs patterns of acoustic realization.

4aSC13. Variability of formant values at different time points of vowels. Jonathan Jibson (Univ. of Wisconsin–Madison, 7134 Helen C White Hall, 600 N Park St., Madison, WI 53711, jibson@wisc.edu)

Vowel inherent spectral change research has led to two apparently contradictory conclusions about the mental representation of vowels: perception studies support a two-target model of vowels, while production studies focus on calculating continuous formant contours. This study explores production data in a new way by analyzing where in vowel duration variability is lowest across tokens. Portions of vowels that are less variable across tokens can be taken as reflecting greater importance to or planning by speakers. Born-and-raised Wisconsinites aged 18–30 were recorded reading a word-in-frame reading list eliciting 30 tokens of 14 American English vowels (all but /ə, ɚ/) in the context h_d. F1 and F2 were sampled at 50 evenly spaced intervals for each vowel, covering the entire duration. For every speaker, the standard deviations of F1 and F2 were taken. Values for F1 were almost entirely below the perceptual threshold (0.28 Bark), while values for F2 were mostly above. The all-speaker-averaged values were subsequently plotted as a pair of bar charts for every vowel. F1 plots are primarily U-shaped with vertices near midpoints, as are half of F2 plots, with the other half being flat. Neither shape supports a two-target representation of vowels from a production standpoint.

4aSC14. Acoustic characteristics of belching in speech. Brooke L. Kidner (Linguist, Univ. of Southern California, 620 McCarthy Way, #273, Los Angeles, CA 90089, bkidner@usc.edu)

"When such nonspeech sounds occur connected in speech they sometimes carry important social connotations (Pike, 1943: 39)." This sentiment is attested over and over again in linguistic literature that provides cases of nontraditional sounds (such as coughing, moans, cries) being co-opted into the continuous stream of speech within discourse (Ohala,1995; Ogden, 2013; Prelock and Hutchins, 2018; Pinto and Vigil, 2018). This reveals an unanswered problem: While many of these sounds have been able to be expressed by their phonetic parameters in paralinguistics (Trager, 1958; Poyatos, 1975), belching—as evident from a case study as being intentionally interjected into the speech stream with social connotations (Kidner, 2018)—has consistently eluded detailed description using acoustic parameters. Acoustic data were taken from this case study, and belching was distinguished in Praat (Boersma and Weenik, 2018) from modal speech by an increase in glottal pulses and jitter (>4.4%), an increase in shimmer (>15%), low pitch (≤300 Hz), and an irregular amplitude contour. These criteria for belch identification (while not exhaustive) allow us not just to classify belches phonetically within the corpus, but moves the process of acoustically defining otherwise difficult paralinguistic sounds forward.

4aSC15. A curvilinear comparison of normalized English formant trajectories. Byunggon Yang (English Education Dept., Pusan National Univ., Pusantaehakro63-2 Keumjunggu, Pusan 46241, South Korea, byunggonyang@gmail.com)

Formant trajectories reflect the continuous variation of speakers' articulatory movements over time. This study examined formant trajectories of English vowels produced by ninety-three American men and women in Hillenbrand *et al.* (1995). *Praat* scripts were used to read the sound data, and to collect formant values at the six equal time points within each vowel segment. The scale function in *R* was used to normalize the first two formant data sets of men and women separately. Then, the scaled trajectories were compared using generalized additive mixed models (GAMMs). The results indicate that women yielded proportionately higher formant values than

men. The standard deviations of each group showed similar patterns at the first formant (F1) and the second formant (F2) axes and at the measurement points. GAMMs of all the scaled formant data produced various patterns of deviation along the measurement points. Generally, more group difference exists in F1 than in F2. Also, women's trajectories appear more dynamic along the vertical and horizontal axes than those of men. We conclude that scaling and nonlinear testing are useful tools for pinpointing differences between speaker group's formant trajectories. This research could be useful as a foundation for future studies comparing curvilinear datasets.

4aSC16. Contextual and speaker variation of glottalic stops in Q'eqchi'. Kevin Liang (Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing, Philadelphia, PA 19104-6228, kevliang@sas.upenn.edu) and Jianjing Kuang (Univ. of Pennsylvania, Philadelphia, PA)

Because of the complex coordination of laryngeal and supralaryngeal articulations, the realization of glottalic stops is highly variable across both languages and speakers. Therefore, it is important to discern the range of variation of glottalic stops. In Q'eqchi', like other Mayan language, the production of bilabial glottalic stops varies between implosive and ejective. The present study explores whether the phonetic realization of glottalic stops in Q'eqchi' systematically varies based on phonetic context. We also investigate whether different speakers have preferences for certain realizations of glottalic stops. Examining acoustic measures, we found that (1) both types of glottalic stops (implosive and ejective) are variably produced by speakers, and they are acoustically distinct, (2) ejective-like stops are more likely to occur in word-initial position, and that, (3) individual speakers have preferences towards either producing more implosive-like stops or more ejective-like stops. This study supports the hypothesis that variation of glottalic stops is conditioned by phonetic context, and that differing speakers have different preferences for the realization of these stops.

4aSC17. Within- and between-speaker acoustic variability: Spontaneous versus read speech. Yoonjeong Lee (UCLA, 31-19 Rehabilitation Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, yoonjeonglee@ucla.edu) and Jody E. Kreiman (UCLA, Los Angeles, CA)

Using principal component analysis (PCA), our previous study [*JASA* 145(pt. 2), 1930, (2019)] of read sentences found surprisingly similar acoustic voice spaces for groups of female and male talkers and for the individuals within groups. Formant frequencies and the balance between higher harmonic amplitudes and inharmonic energy in the voice accounted for the most acoustic variance within and across talkers, but many further details varied idiosyncratically for individual talkers. In this study, we replicated this finding using a set of recorded phone conversations from 99/100 original speakers (49 F), hypothesizing that the same measures would characterize both individual and population acoustic spaces, despite greater acoustic variability for spontaneous utterances. F0, formant frequencies, spectral noise, source spectral shape, and their variability were measured every 5 ms from vowels and approximants. Individual and group PCAs revealed that the acoustic voice spaces derived from spontaneous speech are highly similar to those spaces previously identified based on read speech. One significant difference between the two speaking styles was that unlike read speech, variability in F0 emerged as one of the variables that accounted for significant acoustic variability in spontaneous speech. Implications for voice learning, recognition, and discrimination will be discussed. [Work supported by NIH/NSF.]

4aSC18. Effects of stop voicing on long-distance acoustics. Lisa Lipani (Univ. of Georgia, 142 Gilbert Hall, Athens, GA 30602, llipani@uga.edu)

In Coleman (2003), the phonological feature [voice] was found to have a long-distance coarticulatory effect. However, voicing is not always realized as categorically present or absent. Davidson (2016) classifies partially voiced stops as having different patterns of voicing: bleed, trough, negative VOT, and hump. The present study answers the question: How do voicing patterns in stops affect the long-distance coarticulatory properties of the stop? Following the methodology of Coleman (2003), LPC coefficients, voicing, F0, harmonics-to-noise ratio, and z-score normalized RMS amplitude were measured in the read data from the Rainbow Passage in the Nationwide Speech Project (Clopper and Pisoni, 2006) in order to examine the stops and their coarticulatory effects. In the second sentence of the Rainbow Passage, the phones [b], [d], and [k] and their surrounding acoustics were measured in 10 millisecond frames that were aligned in time using dynamic time warping. A preliminary analysis using data visualization with smoothed conditional means reveals differences in the acoustics. There were several differences in both anticipatory and carryover coarticulation in the first through sixth and ninth through eleventh LPC coefficients. Subtle differences in acoustics as a result of long-distance coarticulation have implications for what makes both human and synthetic speech sound natural.

4aSC19. Voice onset time variation in natural southern speech. Lisa Lipani (Univ. of Georgia, 142 Gilbert Hall, Athens, GA 30602, llipani@uga.edu), Michael Olsen, and Rachel M. Olsen (Univ. of Georgia, Athens, GA)

Sociophonetic research has traditionally emphasized vowels and only recently begun to examine consonants. The emerging literature on consonants has found systematic regional variation in consonant production (Jacewicz *et al*., 2009; Eddington and Turner, 2017). These studies have focused on present-day speakers producing lab-recorded speech and thus provide little historical or naturalistic insight to consonant variation. One way that the production of stop consonants varies is in voice onset time (VOT). In this study, we utilize the Digital Archive of Southern Speech (DASS) (Kretzschmar *et al*., 2013), a collection of 64 sociolinguistic interviews recorded between 1968 and 1983, to explore consonant variation in an unscripted historical setting. DASS was force-aligned using the Montreal Forced Aligner (McAuliffe *et al*., 2017), and the VOT of pre-vocalic, word-initial stop consonants in three-minute audio clips from each speaker was measured using AutoVOT (Keshet *et al*., 2014). VOT was normalized by dividing duration by speaking rate, and this normalized measurement was included as the dependent variable in a mixed effects model. As expected, stress, voicelessness, and a dorsal place of articulation were significant predictors. Our preliminary analysis reveals significant regional and age differences, adding evidence for speaker-specific variation of VOT as a result of sociophonetic variables.

4aSC20. Oral-velum actions in the articulation of nasal juncture geminates and singletons. Miran Oh (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, miranoh@usc.edu) and Dani Byrd (Linguist, Univ. of Southern California, Los Angeles, CA)

Articulatory studies of consonant production generally focus on oral gestures, without further attention to the action of non-oral gestures—such as velum gestures. In large measure, this lacuna has arisen due to a lack of instrumental data. This study deploys real-time MRI video to investigate nasal consonant production in Korean, allowing us to observe spatiotemporal aspects of both oral and velum gestures. We examine the articulatory dynamics of concatenated and assimilated juncture geminate nasals (/n#n/ & /t#n/ → [n.n]) in comparison with singleton nasals (/#n/ & /n#p/). We predict that geminates and singletons differ in the duration of oral and velum gestures and that concatenated and assimilated geminates show different articulatory characteristics. Findings indicate that Tongue Tip (TT) duration is longer in both geminate types relative to singletons, but Velum (VEL) lowering duration does not consistently distinguish geminates from singletons, though its plateau duration does. Further, both TT constriction degree and VEL lowering displacement differ for assimilated geminates as compared to concatenated geminates and singleton codas, with greater TT magnitude and lesser VEL lowering in assimilated geminates. Results suggest that velum and oral components do not behave in parallel in the production of these nasal consonant sequences. [Work supported by NIH.]

4aSC21. The vowel nasalization found in Murcia, Spain: A case study of this phenomenon. Alexandra Lopez Vera (Spanish and Portuguese, Univ. of California Santa Barbara, 793 Willow Walk, Apt. A, Goleta, CA 93117, lopezvera@ucsb.edu)

Of all the linguistic features that characterize the southern dialect spoken in Spain, the vowel nasalization found in Murcia is the most distinctive one. Vowel nasalization in Spanish mostly affects vowels between two nasals and in utterance-initial position following nasals. The nasalized vowels found in this region, however, are not in either of those contexts. As an example, the word *especial* [epeθjal] > [epeθjã] presents a nasalized final vowel in an unexpected context. There are authors believing that the nasalization becomes an acoustic marker of a word limit. The goal of this study is to determine the contexts where nasalization occurs based on the analysis of recordings made by speakers of this dialect. A number of factors were targeted as potential predictors of nasalization, including internal factors such as words with stressed first or final syllable and composed by random consonant phonemes which obey the Sonority Principle: vowel (or diphthong) + /l/, /r/, /d/, /s/ and /ʃ/, and external factors such as gender, age, social class and level of education. Results indicate that women nasalize less than men, age does not have an impact, and the lower the social class or the education is, the higher index of nasalization is found.

4aSC22. Acoustic correlates of Apurímac Quechua ejective stops. Mackenzie Marcinko (Dept. of Linguist and Cognit. Sci., Univ. of Delaware, 125 E. Main St., Newark, DE 19716, marcinko@udel.edu), Jermani Ojeda Ludeña (Univ. of Texas at Austin, Curahuasi, Peru), and Abdulrhman Alshahrani (Dept. of Linguist and Cognit. Sci., Univ. of Delaware, Newark, DE)

Apurímac Quechua ejective and pulmonic stops are compared with respect to *VOT, rise time* (the following vowel's peak amplitude – onset amplitude), and *F0 difference* (following vowel's midpoint amplitude – onset amplitude). These three properties distinguish ejective from pulmonic stops in other languages (e.g., Ingush) but were previously unanalyzed in Quechuan languages. Results show that only rise time and VOT distinguish Apurímac Quechua ejective stops. All syllables (stressed and unstressed) of CV.CV.CV words were analyzed. The vowel /a/ followed each stop onset. 60 pulmonic plus 40 ejective stops were compared (all places of articulation) (α = 0.05). Rise time is the strongest acoustic cue for ejective stops: vowels following ejective stops had significantly longer rise times ($mean_{ejective}$= 7.74 dB, SD = 1.27) than pulmonic ($mean_{pulmonic}$ = 4.85 dB, SD = 2.94). Ejective stops therefore rise to peak amplitude more slowly than pulmonic. F0 difference did not significantly differ between vowels following each stop type ($mean_{ejective}$ = -0.32 Hz, SD = 0.05; $mean_{pulmonic}$ = 0.40 Hz, SD = 0.03). Ejective stops had significantly longer VOTs ($mean_{ejective}$ = 27.82 m/s, SD = 2.01) than pulmonic ($mean_{pulmonic}$ = 23.98 m/s, SD = 4.09). This difference is small, indicating that VOT is a less-robust acoustic cue for ejective stops.

4aSC23. Effects of alcohol and emotions on voice features in Mexican Spanish accented speech: A forensically relevant case study. Viola G. Miglio (Spanish and Portuguese, Univ. of California, Santa Barbara, UCSB, Santa Barbara, CA 93106, miglio@ucsb.edu)

This is a case study of how alcohol intoxication and emotions affect the speech of a male Mexican Spanish speaker arrested on suspicion of vehicular manslaughter, and how his dialectal variety may have influenced the assumption of the arresting officers that alcohol intoxication was revealed by his altered speech patterns. There are still few standards in Forensic Linguistics and Voice Recognition/Identification (cf. the survey by Hollien *et al*., 2014). We do know, however, that listeners often overestimate the amount of alcohol speakers ingested, sober speakers can successfully mimic drunken speech, and conversely it is possible for a speaker under the influence to sound more sober than s/he actually is by modifying certain parameters (*ibid*., p. 180). It has also been demonstrated that dialectal differences can impair intelligibility and speaker recognition (Bahr *et al*. 2002, Betancourt and Bahr, 2010): the present case study analyses the acoustic segmental and suprasegmental aspects of the existing recordings of the arrest, and shows that while the speaker may have been physiologically drunk, he was rather "linguistically sober." Further correlations are also drawn between dialectal characteristics and "slurred speech," as well as the influence of emotions and possibly deception on speech rhythm and variations in vocal parameters.

4aSC24. Shannon-Zipf comparison of human and humpback whale voiced phonetic and sub-unit complexity. Howard S. Pines (Cisco Systems, Retired, Wireless Network Business Unit, 8752 TERRACE Dr., El Cerrito, CA 94530, howardpines@ieee.org)

The striking similarities of time-frequency spectrograms of voiced human speech and humpback whale vocalizations indicated a common targeted, frequency-modulated phonetic/sub-unit basis. To map the sub-unit structure of voiced humpback whale song units, a time-frequency-log-power contour segmentation, extraction, and classification procedure was tested on voiced human speech and then applied to humpback vocalizations. When the extracted "target" tone-pairs of the two most energetic "vocal fold" harmonic frequencies were plotted in x-y coordinates and subjected to cluster analysis, the plot exhibited properties of a Shannon-Hartley-compliant "modem symbol constellation" diagram of 14 distinct tone-pair and single-tone sub-regions. The humpback voiced symbol constellation is structurally comparable to the Peterson-Barney, F1 vs F2, formant mapping of English language vowels. The humpback's core set of 14 fixed-pitch sub-units is augmented by numerous intra- and inter-sub-regional pitch-varying transitions similar to the generation of Asian language tonal vowels. The information entropy and plot of the humpback's complete 65-sub-unit symbol-set's cumulative probability versus 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 units referenced to their constituent sub-unit symbol-codes has enabled the compilation of an "alphabetic-lexicon" of voiced humpback song units.

4aSC25. Quantifying macro-rhythm in English and Spanish: A comparison of tonal rhythm strength in two speech styles. Christine Prechtel (Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095, cprechtel@ucla.edu)

This study quantified macro-rhythm in English and Spanish in two speech styles. Macro-rhythm is defined as phrase-medial tonal rhythm (June 2014), and its strength is determined by the frequency of f0 alternations between peaks and valleys within a phrase, the uniformity of the rise-fall shape, and the regularity of distance intervals between high and low targets. The degree of strength can be predicted based on the number of phrase-level tones in a language's tonal inventory, the most common type of phrase-medial tone, and the frequency of f0 rise per Prosodic Word. Based on these criteria, Spanish is predicted to have stronger macro-rhythm than English: the most common pitch accent is L+<H* in Spanish and H* in English, and Spanish tends to accent content words with greater regularity than English. Two experiments quantified and compared slope shape variability, the number of f0 alternations per Prosodic Word, and the regularity of distance intervals between tonal targets in read speech (Experiment 1) and newscaster speech (Experiment 2). The results of the variability, frequency, and distance measures support the prediction that Spanish has greater macro-rhythm strength than English.

4aSC26. An intonational model of Shanghai Wu: A pitch-accent analysis. Brice D. Roberts (Linguist, UCLA, 10405 Irene St., #305, Los Angeles, CA 90034, Bricedavidroberts@gmail.com)

This paper presents an intonational model of Shanghai Wu in the Autosegmental-Metrical framework. Shanghai's peculiar phrasal tonal system, which spreads lexical tones over sandhi domains, is well-discussed in the literature; however, there is no account of the language's contrastive intonational elements and their phonetic realization. Grounded in phonetic data collected from 14 Shanghainese speakers born between 1945 and 1963, the paper analyzes the language as a pitch-accent system rather than a tonal one, with three distinctive accents: L* + H, H* + L, and L*. These pitch accents pair with one of two accentual phrase-like boundary tones, La and LHa with specific tonotactic requirements. Additionally, I will describe the prosodic hierarchy, as determined by the intonation, as having two levels above the accentual phrase, the intonational phrase (marked via initial pitch

range expansion and one of three boundary tones: L%, %, and H%) as well as the major phrase. Together with a labelling system, this analysis allows for easy annotation of Shanghainese tonal events, similar to the ToBI-type models for languages such as Japanese and Serbo-Croatian.

4aSC27. An acoustic view of the local Thai "Lae" reading and singing. Rungpat -. Roengpitya (Dept. of English, Faculty of Liberal Arts, Mahidol Univ., Thailand, 999 Phuttamonthol 4 Rd., Salaya, Nakhon Pathom 73170, Thailand, rungpat@gmail.com)

This paper explores the acoustic characteristics of the local Thai "Lae" reading, as opposed to the singing one. The so-called "Lae /lǽæ/" with the contents based on the local Thai culture has been the traditional Thai singing for centuries and has been enculturated only within the families of professional vocalists. In this paper, an acoustic study was conducted. Two professional Thai vocalists (one female and one male) were asked to read aloud and sing two Lae songs: the gac song (n = 236 syllables) and the song of the Mahasawat Canal (n = 272 syllables). There were a total of 2032 tokens (508 syllables X 2 speakers X 2 types). All the tokens were digitally recorded in a sound-proof room and acoustically analyzed in the Praat program. The results revealed that, in singing, the vowel duration and the final sonorants (if any) were two-three times longer; the vowels and diphthongs were lower (with higher F1); and the F0 levels for Thai tones were higher than the reading counterparts. Additionally, in vocalization, both female and male vocalists inserted extra syllables and glottalization while singing. It is hoped that this acoustic study will help glocalize and preserve the local Thai Lae singing.

4aSC28. The roles of vowel length and sentential context in onset pitch perturbations in Thai. Alif Silpachai (Appl. Linguist and Technol., Iowa State Univ., 527 Farmhouse Ln., Ames, IA 50011-1054, alif@iastate.edu)

This study investigates the relationship between fundamental frequency at the onset of voicing (onset f_0) and Voice Onset Time (VOT) in a tonal language with prevoiced, short-lag, and long-lag stops. Recent research on Thai and Vietnamese has suggested that higher f_0 in the following vowel is conditioned by long-lag stops, but this effect occurs more in higher, not lower, tones and in words produced in isolation, not in a carrier phrase. An examination of previous studies, however, suggests that the effect may be moderated by vowel length and the type of carrier phrase. To determine whether this is true, this study compares onset f_0 measured 40 ms after voicing onset in Thai low tone words with phonemically short and long vowels that occur in two types of carrier phrases and in isolation. The results show that prevoiced, not short- or long-lag, stops condition higher onset f_0 in short, not long, vowels, and this effect takes place in words occurring in both types of carrier phrases, not in isolation. This suggests that vowel length may be a relevant factor. The results will be discussed further, and implications for onset f_0 control will be offered.

4aSC29. An acoustic study of Iu-Mien tones with a special focus on the role of laryngealization in low tone differentiation. Ela Thurgood (California State Univ., Chico, CA 95929, ethurgood@csuchico.edu)

The interaction of fundamental frequency (f0) and phonation types has been analyzed both in tonal languages in which phonation varies with tone [e.g., Jingpho (Maddieson and Ladefoged, 1985), Green Mong (Andruski and Ratliff, 2000), Jalapa Mazatec (Garellek and Keating, 2011), White Hmong (Esposito 2012)] and in tonal languages in which phonation is not contrastive. Patterns in the distribution of a non-contrastive phonation over tones have also been studied, for example in Mandarin (Keating and Esposito, 2007; Yu, 2010) and Cantonese (Yu, 2010). Previously unnoticed, however, is a phenomenon in which the same phonation type differentiates between two tones exclusively on the basis of differences not in its presence but in its distribution over the duration of the vowel. Here we explore an apparent case of this, investigating how f0 and laryngealization interact in Iu-Mien low tones and how this interaction is manifested across the vowel. The results show that low tones correlate with non-modal phonation. What is more, the study also shows that the distribution of laryngealization across the vowel length separates the low rise tone from the low rise-fall tone despite their pitch contours having been run together. This finding points to the development of contrastive phonation in Iu-Mien.

4aSC30. The interaction between lexical prominence and pre-boundary lengthening in Japanese. Karen Tsai (Linguist, Univ. of California, Santa Barbara, Dept. of Linguist, University of California, Santa Barbara, Santa Barbara, CA 93106, karentsai@ucsb.edu) and Argyro Katsika (Linguist, Univ. of California, Santa Barbara, Santa Barbara, CA)

Pre-boundary lengthening, also known as phrase-final lengthening, is a well-established phenomenon. In stress languages, pre-boundary lengthening has been shown to interact with lexical prominence (i.e., stress), affecting the boundary-adjacent constriction gestures in stress-final words, but being attracted towards the stressed syllable when stress is non-final. However, the interaction between pre-boundary lengthening and lexical prominence in a language with lexical pitch accent, like Japanese, is not well understood. The current study uses kinematic data collected with Electromagnetic Articulography (EMA) to investigate this interaction in Japanese. We examine pre-boundary lengthening in trisyllabic phrase-final words as a factor of pitch accent position (initial-accented, medial-accented, final-accented, and no accent). Preliminary results based on duration measurements of the consonant constriction gestures comprising the phrase-final words show that in words with no pitch accent, pre-boundary lengthening has a progressive effect extending two syllables away from the right edge of the phrase, being strongest in the word-final syllable. If pitch accent is present, pre-boundary lengthening is affected by the former's position. The lengthening effect is limited to the final syllable in initial- and medial-accented words, and disappears in final-accented words. Typological dimensions of the interaction between prominence and boundaries are discussed. [Work supported by NSF.]

4aSC31. Sibilant production and perception in the North American West. Charlotte Vaughn (Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403-1290, cvaughn@uoregon.edu), Michael McAuliffe (Dept. of Linguist, McGill Univ., Montreal, PQ, Canada), and Molly E. Babel (Dept. of Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

This paper explores variability in English /s/-/ʃ/ production and perception across the North American West. Although this dialect region has long been considered monolithic, recent work has begun to explore the phonetic diversity within the region. However, much of this work has investigated vocalic variation, and sibilants have remained relatively underexamined. Given reports of /s/-retraction as a change-in-progress in varieties of English around the world (particularly in /str/ clusters), sibilants are of particular interest. In this study, monolingual participants from Oregon, California, and British Columbia provided single word productions and performed a /s/-/ʃ/ perception task. Preliminary results indicate the prevalence of /str/ retraction in the region, with most retraction in speakers from California, followed by Oregon, followed by British Columbia. Region-internal differences in perception of /s/-/ʃ/ were minimal, though a speaker's /str/ retraction ratio contributed to predicting perception for speakers with smaller distances between /s/ and /ʃ/, consistent with prior work finding more perceptual similarity for speakers for whom two sounds are in an allophonic relationship. Taken together, these results contribute to the description of speech in the western region of North America, as well as add data to ongoing questions about variation in production, perception, and their relationship.

4aSC32. Vowel descriptions beyond the first and second formant frequencies. Hong Zhang (Dept. of Linguist, Univ. of Pennsylvania, 3401-C Walnut St., Ste. 300, C Wing, Philadelphia, PA 19104-6228, zhangho@sas.upenn.edu) and Shuang Liu (Dept. of Speech and Hearing Sci., The Ohio State Univ., Columbus, OH)

Conclusions about variation and change of vowels based on the first (F1) and second (F2) formant frequencies may be challenged by at least two major drawbacks intrinsic to this simple 2-D representation. First, F1 and F2 measured at one or two representative points can capture only limited amount of vowel dynamics. The numeric values of F1 and F2 are also prone to errors due to aperiodicity in speech signal. In this study, we explore

alternative ways for more informative and robust vowel representation. We examine both LPC and MFCC based features using two encoding methods. Frequency domain features are first estimated at each time step for the entire vowel duration. The resulting t feature vectors are then passed sequentially through either a set of t independent autoencoders or a recurrent neural network (RNN), both with k hidden units. In case of the autoencoders, each k-by-t vowel matrix is further projected to a lower dimensional space using PCA. In case of the RNN, the k-dimensional hidden vectors are used for cross-condition comparisons. Compared to the simple F1-F2 measurement, our methods are able to capture further nuanced within-category variations across dialect varieties in TIMIT. Results from LPC and MFCC based representations are also compared.

THURSDAY MORNING, 5 DECEMBER 2019 REGENT, 7:55 A.M. TO 9:40 A.M.

Session 4aSPa

Signal Processing in Acoustics and Animal Bioacoustics: Eco-Active Sonar

Brian G. Ferguson, Cochair
DSTO, Locked Bag 7005, Liverpool, New South Wales 1871, Australia

R. Lee Culver, Cochair
ARL, Penn State University, PO Box 30, State College, Pennsylvania 16801

Chair's Introduction—7:55

Invited Papers

8:00

4aSPa1. Eco-active sonar concept with examples. Brian G. Ferguson (Acoust. Systems, Defence Sci. and Technol., Locked Bag 7005, Liverpool, New South Wales 1871, Australia, Brian.Ferguson@defence.gov.au), R. Lee Culver (Appl. Res. Lab., Penn State Univ., State College, PA), and Eric L. Ferguson (School of Elec. and Information Eng., The Univ. of Sydney, Sydney, New South Wales, Australia)

Eco-active sonar is a term coined to describe high-frequency (>20 kHz) wideband (>20 kHz) active sonars that emit environmentally friendly transmissions without impacting the marine ecosystem. The remarkable biological sonar of dolphins is one example of eco-active sonar. The waveforms of the acoustic pulses projected by Indo-Pacific bottlenose dolphins (*Tursiops aduncus*) are reviewed and the observed fine structure of the bistatic scattering impulse responses of the sea floor, sea surface and scatterers in the water volume are presented. Another example is pulse compression sonar (commonly referred to in defense parlance as *low probability of intercept sonar* or *covert active sonar*) which transmit low source level, wideband coded signals of long duration in contrast to high source level, narrowband short-duration pulses of conventional naval sonars. Matched filtering of the sonar returns reveals the presence of echoes from undersea objects and monostatic reverberation contributions from the sea floor and sea surface. High-frequency, high-resolution sonar meets the need for enhanced situational awareness in the undersea littoral as well as precision navigation and high-resolution mapping of the sea floor. The advent of injection-molded piezocomposite wideband sonar transducers with quality factors approaching unity enable the practical realization of eco-active sonar.

8:20

4aSPa2. The SCAT model of wideband biosonar achieves shape imaging, clutter rejection, and mitigation of pulse-echo ambiguity by frequency hopping. James A. Simmons (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu), Chen Ming, Amaro Tuninetti, and Andrea Simmons (Neurosci., Brown Univ., Providence, RI)

Echolocating big brown bats emit wideband (22–100 kHz) FM sounds and perceive echo delay over the full band by auditory spectrogram correlation. However, echoes must contain the lowest FM frequencies (22–30 kHz) as a condition for delay perception; echoes containing fully the higher 90% of frequencies are rejected as having delay images. For echo spectral interference nulls, bats perceive the delay separation of glint reflections corresponding to the frequency spacing of the nulls, a process of spectral transformation. They also perceive echo phase shifts (0 deg to 180 deg) as corresponding alternations in delay (0 ms to ±15 ms). This sensitivity requires registering phase at the lowest frequencies first and is essential for binaural localization and synthetic-aperture shape reconstruction. In dense clutter, bats alternate their lowest frequencies in successive broadcasts from about 22–25 kHz to about 27–30 kHz for "frequency hopping" to mitigate pulse-echo ambiguity for overlapping echo streams in clutter. The SCAT biosonar model replicates all these critical performances. By initiating echo processing at the lowest frequencies and then working up the FM sweeps to complete the delay estimate, it accommodates phase delay for azimuth, synthetic-aperture imaging, and pulse-echo ambiguity when frequency hopping occurs. [Work supported by ONR.]

8:40

4aSPa3. Frequency hopping by big brown bats is coupled with sonar sound grouping in response to pulse-echo ambiguity. Amaro Tuninetti (Brown Univ., 190 Thayer St., Providence, RI 02912, Amaro_Tuninetti@brown.edu), Andrea Simmons, and James A. Simmons (Brown Univ., Providence, RI)

When flying in clutter, big brown bats actively modify their biosonar broadcasts by successively alternating the lowest frequencies in the first harmonic of their FM broadcasts in a phenomenon called "frequency hopping;" and by grouping broadcasts into "sonar sound groups," with short time intervals within a group and longer intervals between groups. These modifications can minimize pulse-echo ambiguity. We analyzed the relationship between frequency hopping and emission of sonar sound groups while bats flew through chain corridors with different clutter densities and through circular hoop tunnels. In more difficult flight tasks, bats emit more sounds in groups of triplets and quadruplets, often with frequency shifts of 1–6 kHz between each sound in the group, creating not only sound but frequency triplets and quadruplets. Echoes of successive sounds remain highly correlated and not separable by conventional receivers according to which broadcast causes them. The consistent pairing of frequency shifting with sonar sound grouping may reflect an auditory streaming mechanism whereby coupled shifts in both frequency and timing can link streams of received echoes to their corresponding emitted calls to disambiguate overlapping echo streams for closely spaced broadcasts. [Work supported by ONR.]

9:00

4aSPa4. Ambisonics and wideband matched-filter approach for bat biosonar study. Hyeon Lee (Dept. of Mech. Eng., Virginia Tech, 100S Randolph Hall (MC0710), 460 Old Turner St., Blacksburg, VA 24061, hlee777@vt.edu), Michael J. Roan (Dept. of Mech. Eng., Virginia Tech, Blacksburg, VA), Chen Ming, and James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI)

We previously introduced a method of bat echolocation signal measurement and analysis for target angle estimation using a custom-built high-frequency (20–80 kHz) soundfield microphone and spatial audio techniques. The method provided precise angle estimates of multiple targets. This presentation will provide details and results of expanding the processing of these spatial audio signals using a wideband matched-filter. The custom-built ultrasonic soundfield microphone recorded a static bat's echolocation signals directly in front of the bat's head. The measured audio signal in B-format was processed and reproduced for all directions with an increment of 1 deg in azimuth and elevation using Ambisonic audio processing techniques. A matched-filter algorithm was developed and produced wideband ambiguity functions (WAF) of transmitted signals from a high-frequency transmitter or bats. The wideband matched-filter outputs filter responses of the reproduced signals in time domain for all directions. The target estimate accompanying the strongest filter response provides the range, normal velocity, angle, and elevation of the target. This presentation shows the hardware and software development including experimental validation to establish this new method for bat biosonar study.

9:20

4aSPa5. Context knowledge-aided sparse bayesian learning method for bathymetric sonar. Jiajun Shen (Harbin Eng. Univ., Rm. 821, underwater Acoust. Bldg., No. 145, Nantong Ave., Nangang Dist., Harbin, Heilongjiang 150001, China, shenjiajun@hrbeu.edu.cn) and Tian Zhou (Harbin Eng. Univ., Harbin, China)

Sparse bayesian learning has recently become successful in many compressed sensing problems. However, their performance critically relies on the appropriate tuning of numerous hyperparameters, because they directly control the statistical distribution and have a significant impact on the performance of the model is being trained. Compared with manual tuning, grid search, and random search, bayesian optimization is a better strategy for intelligently optimizing the hyperparameters. Regardless, traditional bayesian optimization treats each problem as an independent and black box. In this study, the hyperparameters tuning process is improved further due to the contextual knowledge can be incorporate into problem domain such that is easier to optimize. For the bathymetric sonar, the adjacent observation data have certain internal correlation. That is, it is generally believed that the underwater acoustic environment changes slowly, which can be draw inspiration from the ability and incorporate it into the bayesian optimization framework as additional prior information. This extension could yield efficient optimization, without requiring too many iteration steps to converge. The tank experiments are conduct, and the results demonstrate that the proposed method has lower computational burden.

Session 4aSPb

Signal Processing in Acoustics: Array Signal Processing I

Kay L. Gemba, Chair
MPL/SIO, UCSD, University of California, San Diego, 8820 Shellback Way, Spiess Hall, Room 446, La Jolla, California 92093

Chair's Introduction—9:55

Contributed Papers

10:00

4aSPb1. Multi-dimensional scaling (MDS), Euclidean distance matrices (EDMs), matrix completion and outlier identification in array calibration and beamforming. Paul Hursky (Sonar-synesthetics, 4274 Pilon Point, San Diego, CA 92130, paul.hursky@gmail.com)

The matrix of inter-element distances, a Euclidean Distance Matrix (EDM), is symmetric with zero diagonal. It thus has N(N-1)/2 distinct values, which we know are a function of 3N-6 xyz coordinates. Indeed, the EDM for the elements of a 3-D array has rank 3. As a result, it is possible to reconstruct missing elements of the EDM, using a process called matrix completion. Array calibration (aka array element localization or AEL) is determining the relative 3-D coordinates of all array elements, so that they can be incorporated into various beamforming processes. The eigenvectors of a complete and noise-free EDM immediately yield the array element xyz coordinates, using a method called Multi-Dimensional Scaling (MDS). If an EDM has missing measurements and/or is noisy, its matrix completion can recover a complete and more accurate EDM that then yields better relative xyz coordinates. Alternatively, testing a noisy EDM for its know properties can be used to identify outlier measurements (among the inter-element distances), before AEL is performed. We will present the mathematics underlying EDMs, MDS, and matrix completion, and demonstrate AEL results for incomplete and noisy EDMs, using these concepts.

10:15

4aSPb2. Improved array weighting for short nested arrays. Geoffrey F. Edelmann (U. S. Naval Res. Lab., 4555 Overlook Ave. SW, Code 7145, Washington, DC 20375, edelmann@nrl.navy.mil) and Magnus L. Nordenvad (Swedish Defence Res. Agency (FOI), Stockholm, Sweden)

Often linear acoustic arrays are comprised of shorter nested subcomponent arrays cut for higher frequencies. This reduces costs and complexity by trading-off performance, specifically in resolution, SNR, and grating lobes. Typically each sub-aperture is processed individually, and the results incoherently combined over a frequency bandwidth of interest. Here several methods are shown superior to conventional bearing time records on measured at-sea data. Only methods that can be implemented in real-time and on low size, weight, and power systems are considered. Min and product array processing across the sub-apertures results are shown to have advantages, even though the nulls and grating lobes are not optimally coordinated. Finally, convex optimization methods are used to create array weights, over the entire irregularly spaced array. Stability is enforced via a regularization criteria added to the objective function. [Work Supported by ONR and the Royal Swedish Navy.]

10:30

4aSPb3. Robust direction finding in the fluctuating oceans by complex L1-norm principal component analysis. George Sklivanitis (Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33431, gsklivanitis@fau.edu), Gaultier Real (DGA Naval Systems, Toulon, France), and Dimitris Pados (Florida Atlantic Univ., Boca Raton, FL)

We consider the problem of robust direction-of-arrival (DoA) estimation in dynamic ocean environments. Robust direction finding of underwater acoustic signals transmitted from known locations may enable accurate underwater localization and enhance link communication rate by instructing a receiver to listen for transmissions from a specific direction. We propose to estimate the DoA of underwater acoustic signals via subspace methods, executed at a very large receiver array, that involve performing what is known as principal-component analysis (PCA) for finding the L2-norm principal vector subspace of the recorded signal snapshots. However, in practice coherence loss, which typically arises from dynamic wavefront fluctuations due to internal waves, scattering from the sea surface and/or bottom and other unknown environmental parameters, may result in recorded signal snapshots that are corrupted by faulty measurements, leading to an inaccurate estimation of the DoA and source position. We propose to model the loss of coherence as multiplicative random noise applied to the measured acoustic signal. In such cases, L2-norm PCA methods suffer from significant performance degradation. Motivated by the resistance of novel L1-nrom-derived subspaces against the impact of irregular, highly deviating points in reduced-dimensionality data approximations, we propose to employ L1-norm (absolute error) maximum projection PCA of the antenna array measurements and evaluate the performance of a novel, outlier-resistant DoA estimation algorithm.

10:45

4aSPb4. Experimental validation of a distributed sensor network concept. Peter C. Mignerey (Acoust. Div., Naval Res. Lab., Peter Mignerey code 7160, Washington, DC 20375-5350, peter.mignerey@nrl.navy.mil), Steven I. Finette (Acoust. Div., Naval Res. Lab., Washington, DC), Geoffrey F. Edelmann, Lloyd Emokpae, and Jeffrey Schindall (Acoust. Div., Naval Res. Lab., Washington, DC)

Autonomous distributed systems comprising many sensors provide a new paradigm for passive detection of acoustic sources in the ocean. Such systems can be designed to achieve global consensus concerning source detection through distributed computation without the presence of a fusion center. In this presentation, individual sensors make information-theoretic assessments of whether a target is present or absent, and communicate that information within a network to enable a global detection consensus. To obtain the consensus, each sensor obtains neighboring information that populates a state vector, which is reduced by an inner-product with a dynamical-system iteration vector to obtain a scalar estimate of the global information. The process iterates until the network reaches a consensus on

the detection information, after which the process starts anew on a subsequent block of acoustic data. An experiment was conducted on the New Jersey Shelf in early May 2019 to evaluate the above ideas. This talk will cover an overview of information-theoretic detection, the convergence of a linear state system on a network graph, robustness with respect to node failure, and the behavior of a dynamical detection system deployed in the ocean. [Work supported by the U.S. Office of Naval Research.]

11:00

4aSPb5. Incorporating uncertainty into detection probabilities and receiver-operator characteristic curves. Brian M. Worthmann (Lawrence Livermore National Lab., 7000 East Ave., Livermore, CA 94550, bworthma@umich.edu)

Receiver-operator characteristic (ROC) curves are used in a variety of fields to assess the performance of binary classification algorithms. A variety of metrics can be defined that quantitatively summarize ROC curve performance, including area under the curve (AUC). In measured data, the detection statistic for each trial typically has some uncertainty associated with it. ROC curves generated from limited measured data that ignore this uncertainty can appear stair-stepped, and produce a single value for the AUC, which lacks any estimate of its variance. Incorporating the uncertainty in the detection statistic directly can lead to smooth ROC curves parameterized by a mean and variance, which then lead to AUC's with both a mean and variance. With this, the significance between two or more ROC curves, and their AUC's, can be assessed statistically. If the classification algorithm that produced the detection statistics have tuning parameters, the optimal values for these tuning parameters can also be determined, along with their associated variances. This methodology is presented alongside other techniques from literature. [This work was performed under the auspices of the U.S. Department of Energy by Lawrence Livermore National Laboratory under Contract No. DE-AC52-07NA27344 and is supported by the Office of Naval Research.]

11:15

4aSPb6. Universal adaptive beamforming for moving interferers. John R. Buck (ECE, UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, jbuck@umassd.edu)

Loud moving interferers challenge adaptive beamformers (ABFs) attempting attenuate these interferers while detecting quiet sources. Specifically, moving interferers create a dynamic tension in choosing the best window length for estimating the sample covariance matrix (SCM) from received snapshots. If the window is too short, the ABF does not accurately steer its nulls toward the interferers, diminishing its effectiveness in attenuating the interferers. If the window is too long, the interferers transit multiple beamwidths, splitting their power across multiple eigenvectors, also diminishing the ABF's effectiveness (Baggeroer and Cox, 1999). Previously proposed approaches to address this challenge include Multirate Adaptive Beamforming (Cox, 2000) and Null-Broadening (Song *et al.*, 2003). This talk proposes a universal adaptive beamformer (UABF) exploiting online learning techniques to adapt the length of the SCM averaging window. The UABF's array weights blend the array weights of several competing ABFs. The recent performance of each fixed-window ABF determines its contribution to the UABF array weights. The UABF's performance provably converges to rival the best fixed-length window ABF for every finite power input sequence, without assuming a specific probability distribution. In non-stationary environments where interferers change speed, the UABF may outperform all of the competing fixed-window ABFs. [Supported by ONR 321US.]

11:30

4aSPb7. Automatic detection and tracking of contacts based in clusterization applied in passive sonar. Fabio O. Silva (Instituto de Pesquisas da Marinha, Rua Ipiru, 2, Rio de Janeiro, Rio de Janeiro 21931-090, Brazil, obs.fabio@gmail.com), Fabrício Bozzi, Fernando d. Monteiro, William S. Filho (Instituto de Pesquisas da Marinha, Rio de Janeiro, Brazil), and Carlos F. Soares (Universidade Federal do Rio de Janeiro, Rio de Janeiro, Brazil)

In passive sonar, the detection and tracking of contacts based on the energy of Direction of Arrival (DoA) is a challenge for the spatiotemporal clusterization algorithms. Typically, the tracking of spatiotemporal data involves two steps, first a clusterization of a temporal frame to reduce the data to micro-cluster and a second step tracking the micro-cluster moving. That tracking approach has problems to predict the motion and resolve merge and split of clusters. In this paper, we propose, analyze e test an algorithm to explore the characteristics of passive sonar̉ signals and improve the tracking performance. To resolve the bearing crossing we proposed an automatic parameter extraction on Low-Frequency Analysis and Recording (LOFAR) and Detection of Envelope Modulation on Noise (DEMON) with the same tracking abording to extract the contact acoustic signature and in case of crossing reassociate the contact based on his signature. The proposed algorithm is analyzed in a theoretical dataset to parameters tuning and evaluated in a real dataset collected by a flank array in a navigation channel.

Session 4aUW

Underwater Acoustics: Measurements, Instrumentation, and Analysis Methods

Ilya A. Udovydchenkov, Cochair
Sensors, Electromagnetics, and Electronic Warfare, The MITRE Corporation, 202 Burlington Rd., Bedford, Massachusetts 01730

Benjamin Dzikowicz, Cochair
Naval Research Laboratory, 4555 Overlook Ave. SW, Washington, DC 20375

Contributed Papers

8:15

4aUW1. Deployment of static underwater acoustic recorders in high tidal flow environments. Paul A. Lepper (Wolfson School, Loughborough Univ., Loughborough LE113TU, United Kingdom, p.a.lepper@lboro.ac.uk), Steven LLoyd (Wolfson School, Loughborough Univ., Loughborough, United Kingdom), Steven Benjamins (Scottish Assoc. of Marine Sci., Oban, United Kingdom), and Ben Wilson (Scottish Assoc. of Marine Sci., Oban, Argyll, United Kingdom)

The growth of interest in the development of harnessing tidal energy has also led to the requirement of assessment of underwaters sound fields in these environments. The radiated acoustic signatures of underwater systems in these environments is of significant interest. The measurements of the sound field in tidal flow areas however offer a number of major challenges. In a high flow environment, a static hydrophone would often suffer from significant flow noise. This parasitic noise is not real in the environment but a result of water column particle interaction with the hydrophone surface and has the capability of severely limiting the systems dynamic range for assessment of the true acoustic environment. Drifting system, moving at the speed of the water flow minimise this by reducing fluid particle to sensor interactions however are constrained by both temporal and spatial variations during the drifting limiting the ability to continuously monitor a system. Flow shield technology (sonar domes) have however been used extensively of the reverse problem of high-quality sound reception from a moving platform essentially suffering from the same problem. Work is presented on the use flow shield concept for a proto-type seabed mounted static recorder deployed in a high flow environment.

8:30

4aUW2. Distortion of the ambient low-frequency acoustic field by moorings. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, 833 Dyer Rd., Bldg 232, Monterey, CA 93943-5216, oagodin@nps.edu)

Moored receivers are often employed to collect long time series of underwater acoustic data to monitor the ocean. Moorings typically contain a float and other elements, acoustic parameters of which are rather distinct from those of seawater. In the frequency range of primary interest in seismo-acoustics and noise interferometry, the size of a mooring and dimensions of its float may be of the order of the acoustic wavelength or smaller. Then, acoustic measurements are performed in the near field of a strong scatterer. Assuming a spherical shape of the float, this paper aims to quantify the difference between the ambient and measured characteristics of low-frequency acoustic fields. Diffraction is shown to significantly affect low-frequency acoustic fields in a vicinity of a compact object even when the object is small compared to the wavelength. Scattering-induced low-frequency perturbations in oscillatory velocity are generally much larger than pressure perturbations in the vicinity of sub-wavelength objects. These perturbations prevent mounted vector sensors from measuring correctly the direction of the free-field oscillatory velocity. The feasibility of a posteriori compensation of the distortions in scalar and vector sensor measurements will be discussed. [Work supported by ONR.]

8:45

4aUW3. Estimation of clock errors in underwater acoustic instruments. Ilya A. Udovydchenkov (Sensors, Electromagnetics, and Electron. Warfare, The MITRE Corp., 202 Burlington Rd., Bedford, MA 01730, iudovydchenkov@mitre.org), Ballard J. Blair (Communications, SIGINT & PNT, The MITRE Corp., Bedford, MA), Ralph A. Stephen (RASCON Assoc. LLC, West Falmouth, MA), and Dianne E. Egnor (Communications, SIGINT & PNT, The MITRE Corp., Bedford, MA)

All underwater acoustic sensors require accurate on-board clocks for subsequent data analysis and interpretation. Unfortunately, most clock oscillators change their frequency over time, which occurs due to environmental changes, aging, and other factors. Typically, clock errors are accounted for by referencing the instrument clock to a known accurate clock (such as GPS) before and after the deployment and applying a correction during data post-processing. This method, however, does not accurately estimate clock errors during a specific experimental event. A method to estimate clock performance during the specific event of interest has been developed. It is shown that advanced signal processing techniques can be used to accurately reconstruct the motion of the acoustic source, which, in turn, can improve the accuracy of the acoustic receivers deployed on the seafloor in the deep ocean. A small subset of data collected on the last day of the Ocean Bottom Seismometer Augmentation in the North Pacific (OBSANP) Experiment in June–July 2013 is analyzed for the demonstration of these techniques. [Work supported by the MITRE Corporation's R&D Program.]

9:00

4aUW4. Investigation on measurement method for hydrodynamic noise in centrifugal pumps using reverberation tanks. Yiming Zhang (Harbin Eng. Univ., Harbin 150001, China, zhangyiming1993@hrbeu.edu.cn), Qi Li, Rui Tang, and dajing shang (Harbin Eng. Univ., Harbin City, Heilongjiang Province, China)

Pumps are considered as a primary source of acoustic energy in industrial piping. At present, the measurement of the hydrodynamic noise of the pump is done by installing the sensors in the pipeline attached to the pump. This technique is based on the fact that plane waves are formed inside the pipeline and the pulsating pressure travels in the form of sound waves. However, in actual there are no plane waves in an elastic pipe due to the phenomenon of low frequency cut-off. In fact, this pulsating pressure inside the pipeline will result in further vibration of the pipeline. This paper proposes a method for measuring the hydrodynamic noise of the pump by using the reverberation tank. The vibration of a pipeline attached to a centrifugal pump can be avoided by replacing the elastic pipeline with a flexible hose.

The water outlet of the hose is then placed in the reverberation tank fitted with hydrophone arrays. The hydrodynamic noise of the water pump is measured using low frequency expansion technique of the reverberation method. Experiments show that this method can be used effectively to differentiate between the hydrodynamic noise and other associated noises of a pump.

9:15

4aUW5. Experimental evaluation of passive underwater acoustic markers. Aprameya Satish (Mech. Eng., Georgia Inst. of Technol., Woodruff School of Mech. Eng., 801 Ferst Dr., Atlanta, GA 30313, aprameya.satish@gatech.edu), David Trivett, and Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Autonomous Underwater Vehicle (AUV) navigation requires accurate positioning information underwater which is most commonly achieved using active acoustic transponders. Such transponders typically require maintenance, power and are not covert. As an alternative, the concept of using passive underwater acoustic markers with layered media for underwater navigation has been previously explored theoretically and experimentally using flat interfaces [Satish *et al.*, *JASA-EL* **145**(1), EL84–EL89]. In order to design passive acoustic markers with a more isotropic scattered response, acoustic markers with curved symmetry were investigated experimentally using scaled ultrasonic testing of 3-D printed concentric hemispheres. Subsequently, larger acoustic markers using acrylic hemispherical shells are constructed and insonified by conventional high-frequency SONAR, and their acoustic signatures are measured and compared to theoretical responses. Tag design considerations and preliminary results for the scattered response from different acoustic markers are compared for uniqueness of acoustic signature.

9:30

4aUW6. Analytical models for predicting required response of mandrel-type fiber optic acoustic sensors for ocean and seismic applications. Fred C. DeMetz (President & CTO, Sound Path Technologies, LLC, 9056 Camellia Ct, Ste. 100, Rancho Cucamonga, CA 91737, fred@soundpathtech.com)

Fiber optic acoustic sensor systems have been developed by industry, university, and government research teams which meet enhanced requirements for arrays with high sensor counts, high dynamic range, wide bandwidths, and will operate reliably for a decade or more in high-pressure-high-temperature pipeline, ocean, and seismic environments. Optical sensors do not require vulnerable electrical telemetry in the "wet" end of the system, and can reliably transmit high sensor count string outputs which are 10's of km from the light source and acoustic data processors. This paper introduces a modified version of the SONAR equation which allows a straight forward estimation of the required fiber optic acoustic sensor array response to capture the desired acoustic signal detail necessary to interpret the target or seismic event. This enables system design engineers to define and optimize sensor response requirements early on, and help minimize prototype and production system performance and budget risks.

9:45

4aUW7. Research on creation and evolution of artificial diffuse sound field. Jundong Sun (Acoust. Sci. and Technol. Lab., Harbin Eng. University, No. 145 Nantong St., Nangang District, Harbin, Heilongjiang, China, sunjundong@hrbeu.edu.cn), Qi Li, Dajing Shang, and Rui Tang (Acoust. Sci. and Technol. Lab., Harbin Eng. University, Harbin, China)

In a closed space, it is difficult to create a diffuse sound field by using a single source in the reverberation tank. In this paper, a methodology for construction of diffuse sound field using multiple source excitation is proposed. Proposed methodology is based on space processing technique for a single source sound field and also involves reciprocity of sound field transfer function. Firstly, the analytical method of acoustic excitation reverberation field is studied and subsequently the sound pressure equation for excitation reverberation sound field in case of multi sound is obtained. In addition, the numerical method to observe the influence of sound source distribution upon uniformity of sound field is studied by the acoustic analysis software ACTRAN. Finally, diffuse sound field in a glass tank of 1.5 m $\times$ 0.9 m $\times$ 0.6 m dimensions, is constructed by using 30 emission transducers of the same type. In the end, the diffuse sound field is scanned using scanning array and is then evaluated by measuring the standard difference of the space distribution of sound field. The results show that the standard deviation of the sound pressure distribution in the diffused sound field in comparison with 30 transmitting transducers simultaneously within the reverberation control area is not greater than 3.9 dB under the narrow band condition. Hence, it clearly illustrates that construction of diffused sound field by this method is definitely a viable option.

10:00–10:15 Break

10:15

4aUW8. Experimental progress of noise measurements in reverberation tanks. Rui Tang (Harbin Eng. Univ., No.145, Nantong Str., Nangang Dist., Harbin 150001, China, tangrui@hrbeu.edu.cn), Qi Li, Yiming Zhang, and Xinyue Yu (Harbin Eng. Univ., Harbin, China)

The methods and means for measuring the radiation characteristics of underwater sound sources have been developed under the conditions of reverberant tanks. From the perspective of experimental research, this paper introduces the methods of measuring the radiated sound power of steady-state sound sources and transient sound sources in reverberation tanks. The experimental subjects include the typical transducer under steady-state excitation, the cylindrical shell under steady-state excitation, the transient CW pulse sound signal and the underwater transient spark sound. The measurement principles, experimental system and measurement accuracy involved in each experiment are also listed. The experimental results show that the reverberation tank is an effective experimental device for measuring the acoustic characteristics of underwater sound sources, which has the advantages of simple experimental conditions, high measurement efficiency and strong tolerance for different types of sound sources.

10:30

4aUW9. Measuring the modal shapes of elastic targets using modulated radiation pressure. Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, abawi@hlsresearch.com), Ivars P. Kirsteins (Naval Undersea Warfare Ctr., Newport, RI), Philip L. Marston, and Timothy D. Daniel (Phys., Washington State Univ., Pullman, WA)

In earlier work [T. Daniel *et al.*, *JASA* **140**, 3123 (2016)], we had shown experimentally that the resonance modes of 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 ultrasound beam permits the generation of point-like forces on the target's surface with surgical precision. In addition to exciting target modes with this technique, we show that by physically scanning the surface of the target with the point-like force generated by MRP, the mode shape of the target itself can be determined. The mode shapes are determined from the variations of the amplitude and phase of the radiated sound or the target's surface velocity obtained using a laser vibrometer as a function of force locations. Theory will be developed, and results will be presented from experiments conducted in a tank at Washington State University. In these experiments, the mode shapes of small plates, cylinders, and UXOs were measured with MRP using hydrophone and laser vibrometer measurements and compared against those computed using finite element models.

10:45

4aUW10. Effects of rough surface on frequency responses in non-anechoic tank. Dingding Xie (Harbin Eng. Univ., No. 145, Nantong St., Harbin 150001, China, xiedingding@hrbeu.edu.cn), Qi Li, Rui Tang, and Dajing Shang (Harbin Eng. Univ., Harbin City, Heilongjiang Province, China)

The tank used for acoustical measurements should have sufficiently smooth frequency response curves. However, limited by number of modes, excitation and reception of sound field, the curves in non-anechoic tank fluctuate greatly, which causes unsatisfactory results. In this study, a method that reduces the fluctuation by rough surface is suggested. Numerical simulation by finite element computation is presented for sound field with rough surface. Simulation results reveal that there are more modes when the surface fluctuates randomly. Experiment results in non-anechoic tank with randomly fluctuant surface and in the static one are given. It is shown that smoother frequency response curves can be obtained with stronger modal behavior. The compared results between measurement and simulation show good agreements, which validates the effects of rough surface on frequency responses. This study provides a useful method for measuring the characteristics of narrowband signals in non-anechoic tank, and the applicability of the method is provided.

11:00

4aUW11. A robust and accurate interpolant for tabulated sound speed profiles. John C. Peterson (HLS Res., Inc., 12625 High Bluff Dr., Ste. 211, San Diego, CA 92130-2054, jcp@hlsresearch.com) and Michael B. Porter (HLS Res., Inc., San Diego, CA)

An new interpolation algorithm that is a hybrid of the cubic spline and monotone piecewise cubic Hermite polynomial interpolants is introduced. The resulting algorithm has several properties that make it an attractive choice for the interpolation of sound speed profiles in acoustic propagation models. It has been shown that when interpolants do not have a continuous first and/or second derivative, the resulting acoustic model predictions can exhibit a variety of artifacts such as false or omitted caustics. The first derivative of our interpolant is continuous everywhere. Our interpolant also automatically respects the monotonicity of the underlying data in the sense that it will be monotone (increasing or decreasing) on any interval where the associated data is also monotone. This property avoids the oscillations between data points that can be a source of frustration when using some interpolants such as cubic splines. Finally, the interpolant will also possess a continuous second derivative everywhere, except at those data points where monotonicity considerations take priority. In some cases, the interpolant will coincide exactly with the cubic spline.

11:15

4aUW12. Spiral wave front sonar for active target localization on an unmanned underwater vehicle. Benjamin Dzikowicz (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, ben.dzikowicz@nrl.navy.mil), Jason L. Kost, Jacob T. Heddings (Naval Res. Lab., Washington, DC), and David A. Brown (Univ. of Massachusetts Dartmouth, Fall River, MA)

Spiral wave front sonar is a non-imaging, active sonar technique for remote target localization. It operates by transmitting in rapid succession a leading reference signal, a spiral signal whose phase varies by 2π over the transducer's azimuthal plane, and then a trailing second reference signal. Individual echos can be analyzed for range by time-of-flight, radial velocity by measuring the Doppler effect between the leading and trailing reference signals, and bearing by computing the phase difference between reference and spiral signals across a range of frequencies on a single receive channel. In addition, the spectral response of the target is available for classification algorithms. A prototype compact spiral sonar system (spiral transducer array, hydrophone receiver, amplifiers, and data acquisition) was developed and built for use on a Unmanned Underwater Vehicle (UUV). This talk will focus on analysis of results from initial experiments with this system conducted at NRL's Chesapeake Bay Detachment. Target bearing, range, and radial velocities are computed for a number of targets detected within the field. In addition, several targets were tracked over multiple sonar pings. [Work funded by the Naval Research Laboratory.]

Session 4pAA

Architectural Acoustics and Noise: Architectural Soundscapes II

Gary W. Siebein, Cochair
Siebein Associates, Inc., 625 NW 60th Street, Suite C, Gainesville, Florida 32607

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Keely Siebein, Cochair
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Chair's Introduction—1:45

Invited Papers

1:50

4pAA1. Effects of sound source and its direction on subjects' impression of soundscape in workplace. Takeshi Akita (Dept. of Sci. and Technol. for Future Life, Tokyo Denki Univ., 5 Senju-Asahi-cho Adachi-ku, Tokyo 1208551, Japan, akita@cck.dendai.ac.jp), Naoko Sano, Hanui Yu (Dept. of Sci. and Technol. for Future Life, Tokyo Denki Univ., Tokyo, Japan), and Ayako Matsuo (Graduate School of Adv. Sci. and Technol., Tokyo Denki Univ., Tokyo, Japan)

In the ordinary workplace, there exist many kinds of sound. However, workers usually do not intend to listen to them. To keep soundscape in workplace good, it is necessary to know how they feel such sound that they take no notice of. In the present research, one experiment was carried out for the purpose. Twenty subjects participated in the experiment and experienced three kinds of sound source, that is conversation, music, and signal sound, that had sound pressure level of 50 or 60 dBA and presented from four different directions. Results show that sound of 50 dBA has generally better impression than 60 dBA and signal sound of each SPL is recognized as bothering and not calm. On the other hand, 50 dBA of conversation and music that are presented from behind worried subjects little. Besides, 60 dBA of music presented from right and left side of subjects is felt as obstructive as the 50 dBA of signal sound. As a result, it is suggested that soundscape in workplace should be planned with attention to the person's behavior, meaning and volume of sound source, and its spatial layout.

2:10

4pAA2. The soundscape of Sen. Salgado Filho Avenue, Natal/RNBrazil: The acoustic impact caused by the insertion of semi-exclusive bus lane. Bruna P. Vieira (Dept. of Architecture, Federal Univ. of Rio Grande do Norte, Rua Maria Ivone, Edson Queiroz, 175, Fortaleza, Ceará 60834-472, Brazil, b-pacini@hotmail.com) and Edy J. Barbosa (Dept. of Architecture, Federal Univ. of Rio Grande do Norte, Natal, Rio Grande do Norte, Brazil)

Measures have been adopted in Brazilian roads in order to prioritize collective transportation over the individual. In Natal/RN, the lane next to the sidewalk was destined to the bus circulation at Senador Salgado Filho Avenue, raising questions about the noise impact felt by people at bus stops. Therefore, the objective is to understand the influence of the road configuration on the environmental sound quality to point out elements for mobility plans. The methodology used was: measuring of sound pressure levels at bus stops; questionnaires to verify the evaluation of the pedestrians on the sound impact of the semi-exclusive track; development of acoustic maps based on the current scenario and the reallocation of public bus to the central site. The results demonstrated sound levels on sidewalks higher than those recommended by WHO, and that people are disturbed by the noise due to the proximity of the bus. Through the calibrated models, it was found that the location of the semi-exclusive lane on the right is less noisy than in the central bed, a solution adopted in others roads, showing how the studies of sound impact can be used in urban and mobility planning to increase the pedestrians comfort on sidewalks.

2:30

4pAA3. Contextual design of soundscapes: How to. Fernando J. Elizondo - Garza (Acoust. Lab., FIME-UANL, Calle 8 #422, Col. Villazul, San Nicolas, N.L. 66420, Mexico, fjelizon@hotmail.com)

This paper will be discussed some aspects to be considered in the contextual design of soundscapes in contrast with the classical acoustic design. Aspects as: perception in base of cultural background, perception as a multisensorial integration, attention versus boredom, expectation versus deception, soundscape versus landscape, real versus virtual, deterministic design versus design by compromise, isolation respect other spaces, complying of regulations, etc. Also, the need for an open mind approach in education of soundscape design will be discussed.

2:50

4pAA4. Variations in restaurant soundscapes. Jared A. Paine (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 107 Peter Kiewit Inst., 1110 S 67th St., Omaha, NE 68182-0816, japaine17@gmail.com), Joshua Palakapilly, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

The sound levels and concurrent occupancy at a number of restaurants located in Omaha, Nebraska, have been logged during occupied hours. Information on each restaurant's room shape, materials, seating layout, and ambient noise sources have also been gathered. The time-logged data are analyzed to understand the acoustic environment better, particularly by looking at statistical levels (for example, L10 and L90) and the percent of time that sound levels exceed certain values. Comparisons of the logged sound level data against Rindel's predictive equation for restaurant noise levels which includes occupancy and group size (2019) show good correlation. How different sources of ambient noise and their levels in the restaurant may impact the resulting soundscape is additionally highlighted.

3:10

4pAA5. Multi-unit analysis of noise level exceedances over time in hospitals. Stephanie A. Ahrens (Durham School of Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, Omaha, NE 68182-0816, sahrens@unomaha.edu) and Erica E. Ryherd (Univ. Nebraska Lincoln, Omaha, NE)

The purpose of this project is to expand the knowledge base of health care soundscapes by analyzing noise level exceedances in a relatively large sample of hospitals and units, in order to more clearly define the range of sound levels typically experienced in modern healthcare noise environments. Occurrence rate is a newer measure of how often peak and maximum sound levels occur over a measurement period. Other metrics analyzed in this project include the occurrence rate of minimum sound levels and the difference between maximum and minimum sound levels. These and more traditional sound level metrics are evaluated along with standard percentile noise levels over time. The data analyzed in this project are from the Healthcare Acoustics Research Team (HART) library of hospital research and includes 4 hospitals, 18 units, and several measurement locations in each unit. The large sample size provides a greater degree of certainty in the development of standard levels of noise exceeded over time in hospitals.

3:30–3:45 Break

3:45

4pAA6. Architectural soundscapes. Gary W. Siebein (Siebein Assoc., Inc., 625 NW 60th St., Ste. C, Gainesville, FL 32607, gsiebein@siebeinacoustic.com), Hyun Paek, Marylin Roa, Keely Siebein, Jennifer R. Miller, Matthew Vetterick, and Gary Siebein (Siebein Assoc., Inc., Gainesville, FL)

This paper describes seven elements of soundscape theory that can be applied to the design of architectural spaces including worship spaces, music education spaces, justice facilities, museums, performing arts centers, libraries, restaurants and schools. The paper is focused on the transformative steps that must be taken to translate soundscape data and analysis into the physical form of a building where sound is conceived of as a generator of form and is not necessarily a result of the form. Case studies of recent research and architectural studio classes involved with the design of performing arts facilities will be presented to illustrate how sound can become a part of the conceptual structure of buildings in addition to one of the functions to be accommodated in buildings. The concept of a sonic niche in location, time, level, pitch and form becoming a tangible architectural intervention in a building is explored as one way to translate soundscape theory into architectural space.

Contributed Papers

4:05

4pAA7. The impact of the acoustic environments on the emotional experience of worship spaces. Alaa Algargoosh (Architecture, Univ. of Michigan, 2000 Bonisteel Blvd, Ann Arbor, MI 48109, alaas@umich.edu) and Babak Soleimani (Architecture, Univ. of Michigan, Ann Arbor, MI)

The acoustic properties of a space influence how it is being perceived. However, less is known about the ways in which the acoustic environment elicit different emotional responses. The purpose of this paper is to investigate this question through the study of worship spaces. The Focus on such buildings in this study is due to the essential role of the acoustics in shaping the spiritual experience. Adopting a mixed-method approach, the researchers created virtual experiences of these buildings using 360-deg sound and visual recordings. The recordings were displayed for the subjects through a virtual reality headset. During the experiment, physiological signals of the participants and semantic descriptions of their experience were collected and analyzed to understand the emotional impact of the acoustic environment. The outcome of this study will allow exploring the relationship between acoustic characteristics of worship spaces and the human emotional experience. Therefore, it will provide a further understanding of the impact of the architectural design that determines the acoustic environment, which then contributes to improving the restorative qualities of such spaces and enhancing the spiritual experience.

4:20

4pAA8. Improving the clinical environment soundscape. Ally Taylor (School of Computing, Eng. and Built Environment, Glasgow Caledonian Univ., 3/2, 3, Blackfriars Rd., Glasgow G1 1QG, United Kingdom, ATAYLO206@caledonian.ac.uk), Dr David Moore (School of Computing, Eng. and Built Environment, Glasgow Caledonian Univ., Glasgow, United Kingdom), Liz Simpson, and Wendy Smith (School of Health and Life Sci., Glasgow Caledonian Univ., Glasgow, United Kingdom)

This project examines the impact of the clinical acoustic environment on both patients and staff within a large modern inner-city acute hospital (1677 beds). It aims to both summarize and improve the hospital soundscape. Three key parts of experimental work were carried out: noise level measurements were taken in the Queen Elizabeth University Hospital atrium in Glasgow, interviews were undertaken with medical personnel and listening experiments were conducted to assess if adding natural sounds into the environment could reduce the negative effects noise can have on the occupants of the hospital. The atrium area of the hospital was found to be far louder than needed for rest and relaxation, with all recorded equivalent noise levels over 55 dB(A). Relaxing sounds of nature were added to the hospital soundscape to harness the restorative power of nature. The effect of this was tested using the perceived restorativeness soundscape scale; there was mixed evidence that the introduction of natural sounds led to an increase in the environment's restorative power, with four out of eight attention restoration theory components tested displaying a statistically significant difference. The medical personnel interviewed indicated that the idea of introducing natural sounds is promising and requires further research.

4:35

4pAA9. Distraction factors on doctors and nurses in open-plan examination room. Ainun Nadiroh (Eng. Phys., Inst. Teknologi Sepuluh Nopember, Surabaya, Indonesia), Dhany Arifianto (Dept. of Eng. Phys., Insti. Teknologi Sepuluh Nopember, Sukolilo Campus, Surabaya 60111, Indonesia, dhany@ep.its.ac.id), and Nyilo Purnami (Otorhinolaryngology Head and Neck Surgery, Universitas Airlangga-Dr. Soetomo General Academical Hospital, Surabaya, East Java, Indonesia)

In Indonesia, there exist open-plan examination rooms in government-owned general hospitals serving thousands of patients each day. In this study, we developed a questionnaire with 70 questions to assess the effect of background noise level to 45 doctors-nurses working at pediatric, obstetrics and gynecology, and audiology clinics, respectively at Dr. Soetomo General Academical Hospital. The subjects have been working for more than five years. The background noise includes conversations, baby cries, announcements whose level is about 60 dBA daily. We used a simple statistical technique to determine the most to the least distracting sounds perceptually. The results are, then, crossed the spectrum of the background noises to find out the characteristic of the most intrusive noise. The results showed that the Pediatric clinic has the highest level and dominantly with high pitch baby cries in the waiting lounge. The distraction to either the medical doctor or the nurse may have lost of focus during an examination, determining a diagnosis based on the symptoms, attentively listening to the patient. Furthermore, the intrusive noise characteristic that distracted the staffs were perceptually different. In the on-going research, we investigate a suitable masker to reduce the intrusiveness of the background noises.

4:50–5:10
Panel Discussion

4p THU. PM

Session 4pAB

Animal Bioacoustics, Signal Processing in Acoustics, Acoustical Oceanography, and Computational Acoustics: Applications of Machine Learning to Bioacoustics II

Kaitlin E. Frasier, Cochair
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Kaitlin Palmer, Cochair
Dept. of Computer Science, San Diego State University, 5500 Campanile Dr., San Diego, California 92182

Marie A. Roch, Cochair
Dept. of Computer Science, San Diego State University, 5500 Campanile Dr., San Diego, California 92182

Contributed Papers

1:30

4pAB1. Shannon-Zipf comparison of humpback whale voiced sub-unit complexity for songs recorded near Hawaii and Mexico. Howard S. Pines (Cisco Systems, Retired, Wireless Network Business Unit, 8752 TERRACE Dr., El Cerrito, CA 94530, howardpines@ieee.org)

The striking similarities of time-frequency spectrograms of voiced human speech and humpback whale vocalizations indicated a common targeted, frequency-modulated phonetic/sub-unit basis. To map the sub-unit structure of humpback whale song units, a time-frequency contour segmentation, extraction, and classification procedure was first tested on voiced human speech and then to the analysis of voiced units in humpback whale songs recorded near Hawaii and Socorro Island, Mexico. When the extracted target tone-pairs of the two most energetic "vocal fold" harmonic frequencies were plotted in x-y coordinates, their distribution into three distinct non-overlapping frequency bands suggested the application of three independent K-means clustering sub-tasks. Calculation of the optimum number of clusters spanning each frequency band utilized a technique to determine the value of K which best correlates the K-Means computed, centroid-to-centroid, sub-regional distances with the K-Means-computed mean pitch-transition-vector lengths for all pitch-varying sub-units spanning the sub-regions. The sub-regional cluster mappings exhibited properties of a Shannon-Hartley-compliant "modem symbol constellation" diagram of up to 14 distinct sub-regions. The mappings for the two populations varied in their range of frequencies and distribution of clusters. The Socorro sub-unit's set-size and information entropy are fractional values compared to the Maui set of 65 derived fixed- and variable-pitch sub-units.

1:45

4pAB2. Sub-unit-based classifier boosts unit diversity and information entropy estimates of humpback whale songs. Howard S. Pines (Cisco Systems, Retired, Wireless Network Business Unit, 8752 Terrace Dr., El Cerrito, CA 94530, howardpines@ieee.org)

The ongoing focus of humpback whale song research is the detection and measurement of periodic patterns of discrete units. The expeditious analysis of high volumes of song data has motivated a holistic approach to unit feature-extraction and classification. These whole-unit-feature-approximation and statistical-classification methods generate diminished unit diversity assessments, resulting in reduced Shannon first-order unit-entropy estimates of 4 to 4.5 bits and second-order estimates in the one-bit range. When the feature-extraction and classification of pitch-varying units is referenced to a new sub-unit-based classification model consisting of an "alphabet" of 65 sub-units, the result is a major expansion in the number of unique units compared to previous studies. This precise specification of sub-unit structure has enabled a marked increase in first-order unit-entropy estimates in the 7-bit range, compared to Shannon's 11.8 bit first-order entropy estimate of the dictionary of English language words. The units exhibiting the greatest sub-unit diversity are concentrated in three- to five-minute subsections of songs. When Shannon second-order entropy analysis is restricted to these time-limited sub-sections, the localized entropy estimates increase significantly. Application of the sub-unit-based classifier to the study of fixed-pitch units and additional song data should lead to a further expansion of the "dictionary" of unique units.

2:00

4pAB3. Recognition of marine sound events using an echo state reservoir and a K-nearest neighbor classifier. Cristian E. Graupe (Ocean Eng., Univ. of Rhode Island, 30 Fish Rd., Narragansett, RI 02882, graupec@uri.edu) and Lora J. Van Uffelen (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

A sound event detection and classification algorithm was developed to sift through passive acoustic data, and to extract and classify signals of interest. Segments of data containing interesting sound events were identified from raw audio files using framing techniques and standard deviation-based event detection, and statistical acoustic features from the temporal and spectral domains were extracted for each segment. An adapted cyclical reservoir model was employed to accumulate time varying information over the duration of each sound event, evolving raw features to an echo state space representation with a fading temporal memory. A K-Nearest Neighbor classifier was trained on the echo states extracted from a library of recordings corresponding to broad classes of common ambient marine sound types and self-produced noise from autonomous underwater vehicles. A simple majority voting scheme was then utilized to obtain a single representative class label for each sound event. Validation of the classifier using available underwater acoustic libraries and passive acoustic datasets collected using Seagliders equipped with acoustic sensors indicates small improvements in both the per-frame and per-label performance for a model trained on states over one trained on raw features.

2:15

4pAB4. Learning relationships between bat habitats and biosonar behavior. Michael Goldsworthy (Comput. Sci., Virginia Tech, 155 Otey St., Rm. 323, Blacksburg, VA 24061, michaeljg@vt.edu), Aidan Bradley (Eng. Mech., Virginia Tech, Blacksburg, VA), Xavier Harrison (Creative Technologies, Virginia Tech, Blacksburg, VA), Walter Newsome (Comput. Eng., Virginia Tech, Blacksburg, VA), Georgina Dzikunu (Biological Sci., Winston-Salem State Univ., Winston-Salem, NC), and Rolf Mueller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Echolocating bats are known to adapt their biosonar behaviors according to their surroundings. Vespertilionid bats, for example, adapt their pulse duration and bandwidth as a function of their proximity to foliage. However, these relationships have been based on mostly qualitative observations so far. More powerful methods that are capable of handling large quantitative datasets could produce further and more detailed insights. In the current work, aligned multimodal datasets have been gathered to address this challenge: habitat geometry (laser scans), acoustic scene (biomimetic active sonar system), bat flight trajectories (infrared camera array), and biosonar behavior (microphone array). Machine learning approaches can then be used to analyze relationships between the components of these multimodal datasets. To do this, representations of the bat's surroundings that are suitable for biosonar problems have been developed, for example a probability density function of the angles and distances from the laser scanned points of a habitat model relative to the bat's position and orientation. These representations will serve as inputs to machine learning algorithms that can predict the quantitative features of the bat's biosonar behavior. This research could lead to a better understanding of bat pulse adaptation and also aid in developing biomimetic sonar navigation systems.

2:30

4pAB5. Deep learning of biosonar landmarks for navigation in forest environments. Liujun Zhang (Mech. Eng., Virginia Tech, 1075 life Sci. circle ictas ii, Blacksburg, VA 24060, sdujune@gmail.com), Ananya Bhardwaj (Mech. Eng., Virginia Tech, Blacksburg, VA), Michael Goldsworthy (Comput. Sci., Virginia Tech, Blacksburg, VA), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

Many bat species live in densely vegetated habitats which pose challenges to their biosonar systems. Among these challenges, the problem of identifying prey in clutter has received the most attention. Here, the far less well-studied problem of landmark identification in forest environments has been investigated. To this end, a large data set of about 220 000 foliage echoes has been collected along different tracks located in forested area. The echoes were recorded using a biomimetic sonarhead with flexible noseleaf and pinnae modeled on the periphery of horseshoe bats. Low-dimensional representations of these foliage echoes were created with the encoder portion of a variational autoencoder deep neural network architecture. The feature vectors obtained in this manner were subjected to clustering to determine whether the echo recordings exhibited continuous variability or fell into discernible clusters. The data silhouettes indicate the presence of a small number of distinguishable clusters in the echo data. Furthermore, mapping the assigned cluster labels to the geographical coordinates of the respective echo recordings revealed that different tracks were characterized by a different "fingerprints" of echo classes. Hence, these fingerprints could be a hypothetical basis for biosonar-based navigation in forest environments.

2:45–3:00 Break

3:00

4pAB6. Learning the relationship between biosonar pulse trains and peripheral dynamics in hipposiderid bats. Yanming Liu (Key Lab. of High Efficiency and Clean Mech. Manufacture, School of Mech. Eng., Shandong Univ., 27 Shanda Nanlu, Jinan, Shandong 250100, China, larry.young.ming@gmail.com), Shuxin Zhang (Virginia Tech Int. Lab., School of Phys., Shandong Univ., Jinan, Shandong, China), and Rolf Müller (Mech. Eng., Virginia Tech, Blacksburg, VA)

The biosonar system of hipposiderid bats is characterized by a peripheral dynamics consisting of variable noseleaf and pinna motions as well as temporal variability in the emitted pulse trains. It may be hypothesized that a well-integrated biosonar emission system should coordinate noseleaf and pinna motions with the biosonar pulse trains if this leads to synergistic effects. To investigate whether a relationship between these two sources of variability in the emission system of hipposiderid bats exists, a data set with high-speed stereo video recordings of noseleaf and pinna motions that were synchronized with ultrasonic recordings of the biosonar pulses has been analyzed. Different similarity metrics that were developed for analyzing neural spike sequences have been tested for comparing the individual biosonar pulse trains. Based on these metrics, pulse trains were clustered and the results were compared to the different categories of noseleaf and pinna motions that were found in the video recordings. The results obtained so far for 80 recorded sequences show that noseleaf/pinna motions and pulse trains can be organized into joint clusters, i.e., certain categories of peripheral motions cluster with specific types of biosonar pulse trains. These links could indicate a synergistic relationship between noseleaf/pinna motions and pulse timings.

3:15

4pAB7. Listening to biodiversity under extreme weather. Chia-yun Lee (Biodiversity Res. Ctr., Academia Sinica, Taipei, Taiwan), Tzu-Hao Lin (Dept. of Marine Biodiversity Res., Japan Agency for Marine-Earth Sci. and Technol., Taipei, Taiwan), Chun-Chia Huang (School of Environ. Sci. and Natural Resources, Universiti Kebangsaan Malaysia, Taipei, Taiwan), and Mao-Ning Tuanmu (Biodiversity Res. Ctr., Academia Sinica, 128 Academia Rd., Sec 2, Taipei City 115, Taiwan, mntuanmu@gmail.com)

Climate change is one of the major threats to biodiversity. Besides increasing air temperatures, the magnitude and frequency of extreme weather events have been increasing, and are predicted to increase further in the future. To understand impacts of extreme weather on biodiversity, long-term and frequent data collections on the environment and biodiversity responses are needed. In this study, we used the environmental ultrasounds collected at eight sites across Taiwan to evaluate the usefulness of passive acoustic monitoring for accessing ecological impacts of heavy rains, which are one of the deadliest weather events. We separated acoustic signals of bats, insects and rains using a machine-learning tool we developed. We then investigated temporal changes in the intensity and frequency compositions of bat signals over a heavy rain event. We found significant impacts of the event on the acoustic signals produced by bats. Bat signals took a longer time to recover from the event at the places experiencing heavier and more temporally concentrated rains than at the places with lighter and more spread out rains. This study shows the potential of passive acoustic monitoring for improving our understanding of the responses of biodiversity to extreme weather events under climate change.

3:30

4pAB8. Automatic detection and classification of fish calls in the Northern Gulf of Mexico using energy detectors and a convolutional neural network. Emily Waddell (Marine Biology, Texas A&M Univ. Galveston, 200 Seawolf Parkway, P.O. Box 1675; Bldg 3029 Rm. 130, Galveston, TX 77554, ewaddell@tamu.edu), Kaitlin E. Frasier (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA), John Hildebrand (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA), and Ana Širović (Marine Biology, Texas A&M Univ. Galveston, Galveston, TX)

Passive acoustic monitoring (PAM) is a relatively low-cost way to collect long-term datasets of animal occurrence, which is particularly effective when the animals are not constantly present. However, these long datasets can require substantial effort for manual analysis; therefore, automatic methods are a more effective way to conduct these analyses and extract calls of interest. In this study, acoustic recordings from the northern Gulf of Mexico, about 60 km north of the 2010 Deepwater Horizon oil spill location, collected between July 2010 and May 2017 were analyzed to determine if fish call abundance and diversity has changed after the oil spill. An energy detector was created to extract seven likely fish calls found in these data and a neural network was subsequently applied to classify the detections into different call categories. Daily, monthly, and yearly patterns in calls were investigated and a hypothesis was tested that there was an increase in call abundance and diversity over time, indicating recovery of fish populations

since the oil spill. Such long-term studies demonstrate that PAM can be used as a population assessment tool for fisheries management.

3:45

4pAB9. Machine learning for automated detection of migrating adult eels from ARIS sonar images. Xiaoqin Zang (Energy and Environment Directorate, Pacific Northwest National Lab., PO Box 999 MSIN #K9-33, Richland, WA 99354, xiaoqin.zang@pnnl.gov), Tianzhixi Yin, Zhangshuan Hou, Robert Mueller (Energy and Environment Directorate, Pacific Northwest National Lab., Richland, WA), Paul T. Jacobson (Elec. Power Res. Inst., Glenelg, MD), and Daniel Zhiqun Deng (Energy and Environment Directorate, Pacific Northwest National Lab., Richland, WA)

A machine-learning-based method is developed to detect American eels from ARIS sonar data through several analyses: (1) utilize wavelet transform to filter noises and enhance sonar images; (2) extract individual targets from the training image data with a screening and threshold approach; (3) identify multiple candidate objects from training and testing images using the selective search method; (4) train and apply convolutional neural network models to classify objects into four categories – background, eels, moving sticks, and other objects (e.g., bubble clouds). With laboratory control experiments, favorable conditions for applying the designed detection and classification method are identified. The applicability to the field data, collected at the Iroquois Water Control Dam on the St. Lawrence River, is also evaluated. The machine-learning-based method yields classification accuracy commensurate with human-supervised classification, providing an automated and efficient way to monitor fish migration and passage through hydropower facilities.

4:00

4pAB10. Segmentation of overlapping sources in mixtures of bat echolocation calls. Mohammad Rasool Izadi, Robert L. Stevenson (Univ. of Notre Dame, Notre Dame, IN), and Laura Kloepper (Biology, Saint Mary's College, 262 Sci. Hall, Saint Mary's College, Notre Dame, IN 46556, lkloepper@saintmarys.edu)

Acoustic recordings of animals in their natural habitat often pose complex analysis challenges. For animals in groups, such as bats echolocating in a large swarm, one challenge is analyzing individual sounds that overlap in frequency and/or time with others. We developed a machine learning method, using deep neural networks, that isolates individual sources in the time-frequency domain using only one network that performs in two separate steps; one of detection and the second one of segmentation. In the detection step, the number of sources along with their bounding boxes are estimated. In the segmentation step, masks of individual sources are extracted. This method allows for parsing of individual bat echolocation calls from recordings containing hundreds of overlapping calls so that individual call characteristics can be studied. We will present the methodology and demonstrate the performance from recordings of variable numbers of bats and SNRs.

THURSDAY AFTERNOON, 5 DECEMBER 2019 — EMPRESS, 1:15 P.M. TO 3:00 P.M.

Session 4pAOa

Acoustical Oceanography: Underwater Noise

Likun Zhang, Chair
University of Mississippi, 145 Hill Drive, Oxford, Mississippi 38677

Contributed Papers

1:15

4pAOa1. Underwater noise generated by vehicle traffic in an underwater tunnel. D. Benjamin Reeder (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943, dbreeder@nps.edu), John E. Joseph, and Tarry Rago (Oceanogr., Naval Postgrad. School, Monterey, CA)

Underwater noise measurements were made adjacent to the Monitor-Merrimac Memorial Bridge-Tunnel in the James River, VA during a field experiment in April of 2019. Anticipated sources of underwater noise included wind, waves and associated bubbles, flow noise, boat/ship traffic, industrial shipyard noise and traffic noise radiating from the underwater tunnel. During the field observations, winds, waves/surface conditions, underwater currents, water column structure and tunnel traffic were monitored by anemometers, high-frequency surface radar, ADCPs, echosounders and traffic sensors. The observed noise field exhibited high frequency and tidal-scale variability due to passing surface vessels and tidally driven currents in the estuary; most interestingly, the noise field exhibited variability on a diurnal time scale, closely correlated to the temporal distribution of vehicular traffic in the underwater tunnel. The tunnel traffic underwater noise amplitude varied by 1–5 dB at 10s to 100s of Hertz and exhibited a non-linear relationship to frequency.

1:30

4pAOa2. The ambient noise environment of the Canada Basin during the year 2016–2017. Peter F. Worcester (SIO/UCSD, 9500 Gilman Dr., 0225, La Jolla, CA 92093-0225, pworcester@ucsd.edu) and Matthew Dzieciuch (SIO/UCSD, La Jolla, CA)

The Canada Basin Acoustic Propagation Experiment (CANAPE) was designed as an active underwater acoustic tomography experiment. The large amount of data (Nyquist frequency, 1 kHz) collected over the course of one year from 7 vertical line arrays allows for a unique capability to characterize the ambient sound environment in the deep-water portion of southwestern Beaufort Sea. The data shows a strong seasonal signal as the ice cover forms, grows, and recedes. There is considerable variability from day to day. Furthermore, the vertical arrays allow one to quantify the noise field by angle and depth. The Arctic is a considerably less noisy environment than the temperate ocean and it gets even quieter as the ice thickens. These measurements form an important new base line as the Arctic continues to react to changing climatic conditions.

1:45

4pAOa3. Modelling the coastal, ambient, marine noise field in space, time and frequency. Calder Robinson (Oceanogr., Dalhousie, 1355 Oxford St., PO15000, Halifax, ON B3H 4R2, Canada, Calder.Robinson@dal.ca), David R. Barclay (Oceanogr., Dalhousie, Halifax, NS, Canada), and Svein Vagle (Inst. of Ocean Sci., Sidney, BC, Canada)

The objective of this work is to model the natural ambient noise level in coastal regions based on local environmental forcing and propagation conditions. 546 continuous hours of noise were recorded between April 15 and May 7, 2018 in Sooke Inlet, British Columbia, a coastal, shallow-water region with complex bathymetry and diverse surface traffic. Optimal 1-hour lagged correlation between raw wind speed and hourly minimum sound pressure level (dB re 1 μPa/Hz) is computed as a function of frequency. Low frequencies (10–500 Hz) are characterized by poor correlation due to flow noise and near continuous shipping and small vessel traffic. Mid-frequencies (0.5–10 kHz) show increasing correlation with the shift from ship to wind dominated forcing. High frequencies (10 kHz +) show good correlation with wind. The empirically derived relationship between noise and wind speed is used to validate a predictive ambient noise model, driven by local weather and oceanographic conditions, an acoustic transmission loss model, and a wind to wave energy model. The noise model's sensitivity to temporal and spatial resolution of environmental and forcing data, and knowledge of bathymetry and bottom type will be quantified with the objective of model portability to other coastal regions.

2:00

4pAOa4. Near-surface underwater noise generated by internal solitary waves in the South China Sea. D. Benjamin Reeder (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, SP-311B, Monterey, CA 93943, dbreeder@nps.edu), Yiing Jang Yang (Oceanogr., National Taiwan Univ., Taipei, Taiwan), Steven R. Ramp (Soliton Ocean Services, Carmel Valley, CA), Sen Jan, and Ming-Huei Chang (Oceanogr., National Taiwan Univ., Taipei, Taiwan)

Underwater noise measurements were made over the steep continental slope east of Dongsha Atoll in the South China Sea during a field experiment from 23 May to 25 June, 2019. A single hydrophone was mounted at a depth of 20 m on a mooring deployed at 300 m water depth; temperature/pressure sensors and current meters were also deployed to monitor the water column temperature and velocity structure. Anticipated sources of underwater noise included ship traffic, wind waves and associated near-surface bubbles. Unique to this deep water environment are currents generated by the world's largest observed internal solitary waves (ISWs) which generate noise associated with flow and bubbles subducted to depths in excess of 30 m. The observed noise variability primarily consisted of tidal- and subtidal-scale fluctuations, superimposed upon which were short timescale fluctuations generated by the ISWs. The noise levels observed during typical ISW passage exhibited a non-linear relationship with frequency, increasing by 5–10 dB at mid-frequencies (>1 kHz) and by 10–20 dB at low frequencies (<1 kHz).

2:15

4pAOa5. Experimental study of bubble sources localization in a water tank for oil spill detection. Xing Yang (Dept of Elec. Eng., Univ. of MS, Oxford, MS 38677, xyang4@go.olemiss.edu), Lei Cao (Dept of Elec. Eng., Univ. of MS, Oxford, MS), and Zhiqu Lu (National Ctr. for Physical Acoust., Univ. of MS, University, MS)

Since the Deepwater Horizon oil spill in 2010, there have been increasing demand for a real-time monitoring system that can detect, locate, and assess oil leakages. To this goal, we launched a three-year research project to develop a passive acoustic technique in 2017. The leaked crude oil creates underwater sound through bubble oscillations in an oil spill event. This paper focuses on studying the localization of the source of bubble-induced sounds in an experimental environment. Hydrophone arrays were deployed randomly and used to receive the acoustic signals in a 2 m × 2 m × 2 m water tank. We used spatial spectrum analysis algorithms such as Multiple Signal Classification (MUSIC) and Estimation of Signal Parameters by Rotational Invariance Techniques (ESPRIT) to determine the direction of arrival (DOA) by a hydrophone Uniform Linear Array (ULA), and utilized the multiple 2-D angles to localize the bubbles in 3-D space. We will also investigate the sensitivity of the localization algorithm with respect to the variations of physical features such as the cluster of frequency components and noise level. Experiment results will be presented and discussed.

2:30

4pAOa6. Characterization of sound induced by bubbles released from nozzles. Likun Zhang (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, 145 Hill Dr., Oxford, MS 38677, zhang@olemiss.edu), Xudong Fan (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, Oxford, MS), and Zhiqu Lu (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, University, MS)

In an oil spill event, the leaked crude oil creates underwater sound through bubble oscillations and fluid jets. In this laboratory study, sound induced by bubbles in two types of oil leakage were simulated: a few bubbles and constant flow bubbles. We investigated sound emitted by bubbles released from nozzles, aiming to obtain the characteristics of the oil leakage from the recorded sound signals. For the few bubble case, the dependences of bubble size and frequency on the nozzle diameter were measured. A relation between the average size of bubble and acoustic frequency with the nozzle diameter was used to explain the measurements. In the case of constant flow bubbles, different flow rate was tested. The relation of acoustic energy with the flow rate was obtained from the experimental data. Bubble size distribution was obtained from the spectrum of recorded sound signals. The physical modeling of the bubble sound will be presented and discussed. [Funded by the Gulf Research Program of the National Academy of Sciences.]

2:45

4pAOa7. Using ambient noise to self-localize and time-synchronize independent sensors for localizing gray whales acoustic activity in Laguna San Ignacio, Mexico. 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), Steven Swartz (Laguna San Ignacio Ecosystem Sci. Program, Darnestown, MD), and Jorge Urbán-Ramírez (Departamento de Ciencias Marinas y Costeras, Universidad Autónoma de Baja California Sur, La Paz, Baja California Sur, Mexico)

Beamforming requires array elements whose data is time-synchronized and spacing is known. This typically involves either fixed sensors running off a central acquisition system, or additional equipment, such as pingers or other calibration sources. It has been previously demonstrated that for a diffuse noise field, the time-averaged ambient noise cross-correlation function between array elements can be used to infer both their spacing and relative clock-offset (Sabra *et al.*, 2005). This allows beamforming across fully independent instruments (i.e., from different acquisition systems), whose relative positions are not precisely known, or may vary. Here, we apply this technique to pairs of bottom-mounted recorders deployed between 5 and 15 meters depth with approximately 15 m spacing. The data was collected in Laguna San Ignacio (Mexico) with the goal of localizing gray whale acoustic activity. Results show that even if the noise field is not strictly diffuse nor azimuthally symmetric, the structure of the time-averaged ambient noise cross-correlation function still allows correcting relative clock-offset and drift throughout the deployment. K. G. Sabra, P. Roux, A. M. Thode, G. L. D'Spain, W. S. Hodgkiss, and W. A. Kuperman, "Using ocean ambient noise for array self-localization and self-synchronization," *IEEE J. Oceanic Eng.* 30,338–347 (2005).

Session 4pAOb

Acoustical Oceanography: Underwater Scattering

Derek R. Olson, Chair
Oceanography, Naval Postgraduate School, 833 Dyer Rd., 313b Spanagel Hall, Monterey, California 93943

Contributed Papers

3:15

4pAOb1. Tracking the spatiotemporal variability of the oxic-anoxic interface in the Baltic Sea with broadband acoustics. Elizabeth F. Weidner (Earth Sci., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, eweidner@ccom.unh.edu), Christian Stranne (Geological Sci., Stockholm Univ., Stockholm, Sweden), and Jonas H. Sundberg (Aquatic Resources, Swedish Univ. of Agricultural Sci., Lysekil, Sweden)

The Baltic Sea is characterized by strong stratification, high nutrient loads from anthropogenic sources, and poor circulation. Consequently, an anoxic zone exists in the bottom waters of large regions of the Baltic Proper. Anoxic conditions have led to water column habitat loss, elimination of benthic fauna, disruption of food webs, and altered biogeochemical nutrient cycling. The onset of anoxic zone is concomitant to the depth of the halocline, which separates the fresher, oxygenated surface waters from the deeper anoxic waters. As a result, the interface of oxic-anoxic waters corresponds to a strong impedance contrast brought about primarily by a sharp gradient in salinity. The acoustic reflection from this impedance contrast was tracked utilizing a broadband (45-90 kHz) split-beam echo sounder in the western Baltic Proper. The broadband acoustic dataset provides the means to remotely observe the spatiotemporal variations of the oxic-anoxic interface. In this study, we capitalize on this to discern the mechanisms influencing the vertical distribution of oxygen in the water column and its effects on fine-scale fish distribution. The methodological development of high resolution anoxic layer tracking with the concurrent study of oceanographic and biological features using broadband acoustics opens up a new level of understanding of the fine-scale ecosystem interactions in the Baltic Sea.

3:30

4pAOb2. Measurement of acoustic backscattering from rocky outcrops in Monterey Bay. Jen Gruber (Oceanogr., Naval Postgrad. School, 1 University Circle, Monterey, CA 93943, jenillee.pajewski@gmail.com) and Derek R. Olson (Oceanogr., Naval Postgrad. School, Monterey, CA)

Acoustic scattering is an interaction between acoustic waves and inhomogeneities (interface or volume) which results in acoustic energy being dispersed in all directions. Scattering strength, a measure of the incoherent scattered energy, depends on grazing angle, frequency, bottom roughness, and/or sediment type and can be a source of false alarms for target detection systems. Understanding acoustic backscatter relationships of the sea floor allows for remote sensing applications, as well as target detection performance, in these areas. Acoustic backscatter from sand and mud bottoms has been studied extensively, but only six rocky bottom studies have been performed to date, motivating our study of high frequency acoustic scattering from rocky environments. For this project, acoustic data were collected and analyzed to measure the acoustic backscattering strength of rocky seafloors as a function of grazing angle, bottom roughness, and frequency in Monterey Bay, CA. High-frequency acoustic surveys with two narrowband split-beam echosounders, 200 kHz and 420 kHz, were conducted. Backscatter measurements were compared to existing bathymetric and geologic surveys from California State University, Monterey Bay and the California Geologic Survey to qualitatively compare backscattering strength to roughness features and geology.

3:45

4pAOb3. Measurements of high-frequency acoustic scattering from sea ice over one season in the Chukchi Sea. Anthony P. Lyons (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, University of New Hampshire, Durham, NH 03824, anthony.lyons@ccom.unh.edu) and Gregory Deemer (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, Durham, NH)

Understanding recent and rapid changes in Arctic sea ice calls for extensive and continuous observations of the ice cover. In-situ stations are spatially limited and satellite remote sensing techniques miss key variables of internal sea-ice structure. Utilizing a dataset obtained with 38, 70, and 200 kHz upward-looking transducers in the northeast Chukchi Sea, we take a first look at how acoustic scattering changes in response to environmental forcing. A complete annual cycle of narrow band scattering returns from the winter of 2017–2018 are investigated with comparisons to the overlying sea ice cover. The initial effort in this research is constrained to the stable growth phase of sea ice in winter through the early melt onset the following spring. Individual, minimally deformed ice floes are identified and tracked when located over the deployment sites. Additionally, atmospheric conditions relevant to changes in the internal properties of sea ice are cross-compared to scattering returns. Results from this preliminary analysis will guide pending Arctic fieldwork and the future development of inverse methods for acoustic remote sensing of sea ice.

4:00

4pAOb4. Estimation and validation of sediment properties by normal-incidence echosounder. Lucas Shearer (The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, lucas.r.shearer@utexas.edu), Aaron Gunderson, and Marcia J. Isakson (The Univ. of Texas at Austin, Austin, TX)

Broad implementation of normal-incidence echo sounding has generated a need for sediment characterization models. These models may utilize the magnitude, phase, and/or envelope shape of reflected acoustic waveforms to infer physical, geoacoustic, and engineering properties of ocean sediment. Normal-incidence echo soundings enable the remote survey of vast underwater areas at a fraction of the cost of ground truth sampling methods, i.e., cores and grabs. In collaboration with the Dauphin Island Sea Lab, acoustic data was collected in the Petit Bois Pass off of the Alabama coast. The magnitude, pulse width, and skew ratio of the matched filter output were measured and quantified as bottom loss, full-width at half-max, and the ratio of half-width at half-max, respectively. Bottom loss measurements were used to determine the index of impedance which, through the application of empirically based regression equations, was then used to calculate sound speed ratio, density, mean grain size, and porosity of the sediment along the survey track. The estimated values of sediment properties are generally in agreement with those measured in the laboratory by ground truth survey. [Work supported by the Office of Naval Research, Ocean Acoustics.]

4:15

4pAOb5. Scattering statistics of a negatively buoyant thermal plume. Timothy B. Forge (Oceanogr., Naval Postgrad. School, Monterey, CA) and Derek R. Olson (Oceanogr. Dept., Naval Postgrad. School, Monterey, CA 93943, olson.derek.r@gmail.com)

High frequency acoustic scattering from inhomogeneous ocean environments is often assumed to be entirely coherent or incoherent. We hypothesize that the mean background stratification (which can also be interpreted as long-wavelength components of the inhomogeneity spectrum) cannot be ignored when estimating the total scattered power, leading to the sum of a coherent and incoherent component in the received pressure. In this work, we report laboratory measurements of scattering from a negatively buoyant thermal plume at 70, 120, and 200 kHz. Simple estimation of the coherent component was not possible because the sonar system rectified the signal. An alternative method to estimate the coherent component was to fit the pdf of the scattered field to a Rice distribution. It was found that at all frequencies, the pdf of the field scattered by plume boundary was lighter than Rayleigh and well fit by the Rice distribution. The coherent-to-incoherent power ratio of the field was highest for the lowest frequency. Scattering from the plume center was well fit by a Rayleigh distribution, indicating that the effect of background stratification is minimal away from the plume boundaries. Accompanying temperature and Doppler velocity measurements of the plume were used to support the acoustic measurements.

4:30

4pAOb6. A time-domain model of high-frequency backscatter from sea ice with applications to upward looking sonar data analysis. Anatoliy Ivakin (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th, Seattle, WA 98105, aniv@uw.edu)

Upward looking sonar (ULS) systems are commonly used in ice-covered environments to evaluate sea-ice draft and related characteristics (such as the ice thickness distribution). The involved backscatter data inversion algorithms are mostly based on estimating the two-way propagation time and using some simplifying assumptions which may result in significant limitations, uncertainties, and errors. This work is aimed to develop an improved remote sensing technique of sea-ice characterization based on a more detailed, physics-based analysis of the backscatter intensity time series, the echo shape. The underlying model accounts for two major mechanisms of high-frequency acoustic backscatter, the ice-water interface roughness and volume heterogeneity of the ice layer, provides an estimate of their relative contributions, and includes the echo parameterization in terms of ULS system characteristics (such as the frequency, transmitted pulse' duration and shape, and the directivity pattern). Using this model, computer simulations were performed for the ULS received signal time series to provide a sensitivity analysis for various physical properties of the ice. Optimal parameters of the system to enhance the ice-properties' inversion are suggested which, in particular, may result in a possibility to remotely discriminate between first-year and multi-year ice types. [Work supported by ONR.]

Session 4pBAa

Biomedical Acoustics: Biomedical Acoustics II

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Contributed Papers

1:15

4pBAa1. Site-specific characterization of transmission efficiency of broadband transcranial focused ultrasound. Collin Smith (Univ. of Minnesota, 200 Union St. SE, St. Paul, MN 55455, smit8844@umn.edu), Christopher O'Driscoll, and Emad S. Ebbini (Univ. of Minnesota, Minneapolis, MN)

The efficacy of transcranial focused ultrasound (tFUS) is largely limited by the acoustic properties of the skull. The spatial and spectral variability of acoustic transmission coefficients were measured in three Sprague-Dawley rat skulls. Target areas such as the ventral posterolateral nuclei (VPL) and hippocampus were included in the analysis. The skull samples were interrogated using two systems: a standard, matched transducer setup for transmission measurements with a narrow beam width and a focused array transducer with planar hydrophone transmission measurements. Both systems were designed to operate in the 2 to 4.5 MHz frequency range. Results have shown that the variability in transmission efficiency for different target locations within the same skull varied by 6 dB. Furthermore, measured transmission efficiencies varied by 7 dB on average as a function of frequency in the 2 to 4.5 MHz range in a nonmonotonic manner. Finally, transmission efficiency for the same target window varied by 25% across different skull samples. These results suggest the need for site-specific optimization of the spectral content of tFUS beams taken the skull transmission characteristics into consideration. The results also serve as a basis for validating reflection mode measurements for characterizing transmission efficiency of tFUS beams.

1:30

4pBAa2. Microelectrode arrays (MEA) for measuring spatio-temporal neural activity generated by low-energy focused ultrasound (LEFUS) in *ex vivo* brain slices. Ivan M. Suarez Castellanos (INSERM - LabTAU U1032, 3 Rue de l'Effort, Lyon 69007, France, ivan.suarez@inserm.fr), Apoutou N'Djin, Jérémy Vion-Bailly (INSERM - LabTAU U1032, Lyon, France), Elena Dossi (Sorbonne Univ., AP-HP, La Pitié-Salpêtrière Hospital, Neuroglial Interactions in Cerebral PhysioPathol., Ctr. for Interdisciplinary Res. in Biology, Collège de France, CNR UMR 7241, INSERM U1050, Paris, France), Alexandre Carpentier (Sorbonne Univ., AP-HP, La Pitié-Salpêtrière Hospital, CarThera Res. Team, Paris, France), Gilles Huberfeld (Sorbonne Univ., AP-HP, La Pitié-Salpêtrière Hospital, Neuroglial Interactions in Cerebral PhysioPathol., Ctr. for Interdisciplinary Res. in Biology, Collège de France, CNR UMR 7241, INSERM U1050, Paris, France), and Jean-Yves Chapelon (INSERM - LabTAU U1032, Lyon, France)

The objective of this project was to study the electrical activities elicited by Low-Energy Focused Ultrasound (LEFUS) neurostimulation in *ex vivo* mouse hippocampal brain slices using a MicroElectrode Array (MEA) system. A mixed LEFUS/MEA platform was developed for spatial-temporal recording of neural responses induced by LEFUS exposures in *ex-vivo* hippocampal brain slices from a mouse model. Hippocampal slices were maintained functional in the 60 electrode MEA chip by perfusion with artificial cerebrospinal fluid. The LEFUS system consisted of a 1.78-MHz focused transducer (diameter: 15 mm, radius of curvature: 15 mm). LEFUS pulses (1.1 MPa, 284–568 cycles) were applied on slices at pulse repetition frequencies in the range of 0.2–1 Hz. Recorded signals resulting from LEFUS stimulation were characterized by negative deflections corresponding to LEFUS artifacts, followed by positive deflections corresponding to electrophysiological responses several milliseconds afterwards. This setup thus allows for the recording and differentiation of fiber volleys (latency $\approx$ 1–5 ms) and post-synaptic potentials (amplitude $\approx$ 100–800 μV) as a result of LEFUS stimulation. Such responses were observed in several electrodes across the MEA matrix while exhibiting a progressive drop in amplitude after every applied pulse (Long-term depression). [This project was supported by the French National Research Agency (ANR-16-TERC-0017) the LabEx DevWeCan, and the Focused Ultrasound Foundation (Centers of Excellence).]

1:45

4pBAa3. Whole-cell patch clamp for monitoring low–energy focused ultrasound (LEFUS)-induced electrical activity in individual neurons. Ivan M. Suarez Castellanos (INSERM - LabTAU U1032, 3 Rue de l'Effort, Lyon, 69007, France, ivan.suarez@inserm.fr), Magali Perier, Jérémy Vion-Bailly, Alain Birer, and Apoutou N'Djin (INSERM - LabTAU U1032, Lyon, France)

The objective of this project was to integrate a LEFUS-system to a patch-clamp platform for exploration of LEFUS-modulated and/or -stimulated electrophysiological responses from individual cultured neurons. Mouse primary neurons and human neural progenitor cells were cultured and plated onto 35-mm diameter Petri dishes. A whole-cell patch-clamp setup in current-clamp mode was used to record electrophysiological activity generated by the neurons. The LEFUS system consisted of a 2.2-MHz planar transducer (diameter: 10-mm), administering 0.5–10 ms pulses at pressures of 6–50 kPa. Neuromodulation studies consisted in measuring changes in the activation threshold to electrical stimulation (current density required to trigger an action potential: AP) in LEFUS-treated neurons. Neurostimulation experiments consisted in measuring electrophysiological responses to LEFUS exposures in the form of triggered APs or electrical discharges from the patched neuron. Neuromodulation results showed that the activation threshold required for triggering APs could be either elevated or lowered by approximately 25% following LEFUS treatment. In neurostimulation experiments, LEFUS pressures of 25 kPa and pulse durations of 10 ms were sufficient for either disturbing the membrane potential of the cell and, occasionally, depolarizing the neuron by approximately 70 mV. [This project was supported by the French National Research Agency (ANR-16-TERC-0017), LabEx DevWeCan, and Focused Ultrasound Foundation (Centers of Excellence).]

2:00

4pBAa4. Acoustically stressing biofilms formed by *Pseudomonas aeruginosa*. Lakshmi Deepika Bharatula (School of Chemical and Biomedical Eng., Nanyang Technol. Univ., Singapore 637459, Singapore, LAKSHMID001@e.ntu.edu.sg), Scott Rice (Singapore Ctr. for Environ. Life Sci. Eng., Nanyang Technol. Univ., Singapore, Singapore), Enrico Marsili (Dept. of Chemical and Mater. Eng., Nazarbayev Univ., Singapore, Singapore), and James J. Kwan (School of Chemical and Biomedical Eng., Nanyang Technol. Univ., Singapore, Singapore)

Bacterial biofilms, the complex and dynamic assemblages of bacterial cells, have shown increased tolerance to antimicrobials compared to their planktonic counterparts. Altered metabolism and micro-environments, in addition to the self-generated extracellular polymeric substances (EPS), collectively help to protect the bacteria from the effect of antimicrobials, thereby increasing the necessity for novel treatment methods that are effective against biofilms. To address this, we have investigated the effect of high intensity focused ultrasound (HIFU) in killing or dispersing biofilms. HIFU reduces the total biofilm biomass without significant cell killing. To investigate the molecular mechanism of action of HIFU, we quantified the intracellular concentrations of cyclic di-GMP, which regulates the switch between biofilm and planktonic life-styles. Biofilms grown on polymeric sheets were exposed to HIFU at 0.5 MHz frequency and the c di-GMP concentration was characterized using confocal microscopy. Changes in the c di-GMP activity localized at the acoustic focus were observed as the biomass decreased. The mechanisms promoting these changes were further investigated by examining the shape of individual bacterial colony grown on agar plates. The observed effects were compared to sham (negative control) and c di-GMP overproducing mutant (positive control).

2:15

4pBAa5. Anatomically realistic simulation framework for ultrasound localization microscopy. Hatim Belgharbi (Polytechnique Montréal, 4030 13e av, Laval, PQ H7R 3B1, Canada, hatim.belgharbi@polymtl.ca), Jonathan Porée, Rafat Damseh, Patrick Delafontaine-Martel (Polytechnique Montréal, Montreal, PQ, Canada), Frédéric Lesage, and Jean Provost (Polytechnique Montréal/Montreal Heart Inst., Montreal, PQ, Canada)

Ultrasound Localization Microscopy (ULM) can map vessels at the capillary scale (<10 μm) by acquiring tens of thousands of images in which injected microbubbles are detected and tracked to generate super-resolved blood vessel maps. However, to our knowledge, there are no validation frameworks for ULM image formation algorithms. Herein, we developed a 3-D, anatomically realistic simulation tool based on serial two-photon microscopy (STPM). Rodent brain vasculature was segmented from STPM [1] into a graph-based model, which was then used to generate microbubble flow trajectories according to a velocity-diameter relationship derived from *in vivo* ULM data [2]. Ultrasound signals were obtained using a fast GPU-based simulator and reconstructed using our ULM image formation algorithm. We then proceeded to a parametric study. We first compared 2-D with 3-D imaging. Despite the latter's lower contrast, the gain in spatial information led to a more complete vasculature map. Localization precision was also enhanced since it was freed from the intrinsic localization error in the elevation direction. Also, increasing the number of transmits enhanced localization precision. From this study, we could establish a relationship between microbubble concentration and acquisition time for full network filling and optimal localization. This simulation tool provides thus a ground truth for the validation of ULM image formation algorithms and enables testing of new hypotheses. [1] R. Damseh *et al.*, *IEEE JBHI*(2018). [2] V. Hingot *et al.*, *Sci Rep.* (2019).

2:30

4pBAa6. Brain-wide pulsatility mapping with gated ultrasound localization microscopy *in vivo*. Chloé Bourquin (Biomedical Eng., Polytechnique Montreal, 2900 Edouard Montpetit Blvd, Montreal, PQ H3T 1J4, Canada, chloe.bourquin@polymtl.ca), Jonathan Porée (Biomedical Eng., Polytechnique Montreal, Montreal, PQ, Canada), Frédéric Lesage, and Jean Provost (Biomedical Eng., Polytechnique Montreal, Montreal Heart Inst., Montreal, PQ, Canada)

Cardiovascular diseases are associated with cognitive impairment. Aging arteries increased stiffness leads to an increased pulsatility in downstream vessels and brain structural damage. Hence, brain-wide pulsatility maps could yield a powerful biomarker for neurodegenerative diseases. Ultrasound Localization Microscopy (ULM) can probe the rodent brain smallest vessels, but is currently limited to averaged velocities over multiple cardiac cycles. The objective here was to map the pulsatility using ULM in a rat brain *in vivo* using a cardiac-gated approach. Rat brains were imaged following craniotomy with a Vantage system (L22-14, Verasonics, WA). ECG-gated frames were acquired in groups of 400 (1000 fps) during 8 min following a microbubble injection. More than 20×10^6 bubbles were detected and tracked to map local velocities. Significantly distinct velocities could be measured during systole and diastole in the entire rat brain. Pulsatility indexes from 0.15 in cortical branches to 0.30 in the anterior cerebral artery could be measured, with an increase in blood velocities of 8% to 15% during systole compared to diastole. To our knowledge, this study reports for the first time the mapping of small vessels pulsatility in the entire rodent brain. [We acknowledge the support of IVADO, TransMedTech and Apogée/CFREF.]

2:45–3:00 Break

3:00

4pBAa7. Catheter-mounted dual-frequency ultrasound transducers for intravascular contrast-enhanced superharmonic imaging. Jinwook Kim (Biomedical Eng., The Univ. of North Carolina at Chapel Hill, 116 Manning Dr., 9018 Mary Ellen Jones Bldg., CB7575, Chapel Hill, NC 27599, jinwookk@email.unc.edu), Sandeep Kasoji (Triangle BioTechnol., Inc., Chapel Hill, NC), Eric Markley (Biomedical Eng., The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), xiaoning jiang (Mech. and Aerosp. Eng., North Carolina State Univ., Raleigh, NC), and Paul Dayton (Biomedical Eng., The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Vulnerable atherosclerotic plaques are a source of morbidity and mortality and are hypothesized to be correlated with abnormal development of microvasculature in arterial walls. Thus, there is an interest in assessing microvascularity for the plaque diagnosis. However, conventional catheter-delivered, high-frequency (>20 MHz) intravascular ultrasound imaging is unable to resolve vasa vasorum microvessels. We present here the development of a custom dual-frequency transducer-tipped catheter for contrast-enhanced superharmonic imaging of plaque-associated microvasculature. We exploit our previous design of the miniaturized dual-element, stacked-type needle transducer. A 4/22 MHz dual-frequency transducer with a lateral dimension of 800 μm is assembled with a 70 cm-long drive shaft inside a 4 Fr-catheter. We designed a 4 MHz-transmitter element to generate moderate amplitude (>1 MPa peak-negative-pressure) pulses for bubble excitation. A confocally aligned 22 MHz-receiver element is designed to detect contrast generated superharmonics (>10 MHz) with substantial tissue rejection. We tested the catheter in a tissue-mimicking phantom simulating an arterial lumen with surrounding microvasculature. The *in vitro* imaging data were acquired through a rotational pull-back (2 mm) imaging procedure with microbubble infusion. The resulting superharmonic images display a clearly defined microvessel with an approximated contrast-to-tissue ratio of 12 dB and an axial resolution of 300 μm.

3:15

4pBAa8. Modeling lipid-encapsulated microbubbles using transient network theory. Bashir M. Alnajar (Dept. of Mech. and Aerosp. Eng., Univ. of Colorado Springs, Colorado Springs, CO), Shankar L. Sridhar, Mark Borden, Franck Vernerey (Dept. of Mech. Eng., Univ. of Colorado Boulder, Boulder, CO), and Michael L. Calvisi (Dept. of Mech. and Aerosp. Eng., Univ. of Colorado Springs, 1420 Austin Bluffs Parkway, Colorado Springs, CO 80918, mcalvisi@uccs.edu)

Encapsulated microbubbles (EMBs) are widely used to enhance contrast in ultrasound sonography and are finding increasing use in biomedical therapies such as drug/gene delivery and tissue ablation. EMBs consist of a gas core surrounded by a stabilizing shell made of various materials, including polymers, lipids, and proteins. Lipid-coated EMBs present a unique challenge for modeling due to their relatively large oscillations and nonlinear, viscoelastic properties. We propose a novel model for a lipid-coated, spherical EMB that utilizes a statistically based continuum theory based on transient networks to simulate the encapsulating material. The use of transient network theory permits the viscoelastic properties of the encapsulation—such as stress, elastic energy and entropy—to be calculated locally based on the configuration of lipid molecules. The model requires a minimum number of parameters that include the lipid concentration, and the rates of attachment and detachment of lipids to and from the network. The model closely reproduces the experimentally measured radial response of an ultrasonically driven, lipid-coated microbubble. The model also reproduces experimentally observed nonlinear behavior, such as compression and expansion-dominated oscillations. Furthermore, the model can be readily extended to large nonspherical EMB deformations, which are important in many biomedical applications.

3:30

4pBAa9. Optical coherence elastography for evaluating Rayleigh wave propagation in soft matter-micellar fluids. Hsiao-Chuan Liu (Mayo Clinic, 200 First St. SW, Rochester, MN 55905, liu.hsiao-chuan@mayo.edu), Piotr Kijanka, and Matthew W. Urban (Mayo Clinic, Rochester, MN)

Wormlike micellar (WM) fluids exhibit a self-assembly property and viscoelastic behavior which make them a promising material for representing biological fluids, ultrasoft tissue phantoms and drug delivery carriers. Although WM biochemistry has been investigated, the mechanical properties have been rarely discussed because of the complexity of soft matter systems. Here, we demonstrate that optical coherence elastography (OCE) is capable of measuring Rayleigh wave of micellar fluids. Three solutions with concentrations of 100, 200, 300 and 400 mM, were considered in the study. Ultrasound excitation to produce Rayleigh waves was accomplished using 7.5 MHz burst excitation signals of length 4.0 ms repeated at a rate of 19 Hz. The optical coherence tomography (OCT) acquisition was set to 50 lateral positions and 4000 A-scans for each lateral position at a 76 kHz scan rate for total acquisition time of 52.6 ms. Ultrasound-based shear wave elastography, implemented on a Verasonics system, was used to obtain group and phase velocity of the shear wave in all micellar fluids as reference. Our research results demonstrated that Rayleigh wave in micellar fluids can be evaluated by OCE and mechanical properties like Young's modulus can be assessed. A wide of concentration ranges of WM fluids would provide a more realistic model to mimic biological fluids and tissues.

3:45

4pBAa10. Temperature-dependent dispersion characteristics of Rayleigh waves in soft tissues. Tae-Woo Han (School of Mech. Eng., Yonsei Univ., 50, Yonsei-ro, Seodaemun-gu, Seoul 03722, South Korea, taewoo.han@yonsei.ac.kr), Hunki Lee, Seonghun Im, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., Seoul, South Korea), and Won-Joong Kim (Lutronic, Goyang-si, Gyeonggi-do, South Korea)

The frequency-dependent shear modulus of a soft tissue gives rise to dispersion of (otherwise nondispersive) Rayleigh waves, the measurement of which can offer a means to study various surface properties of the tissue. Among the factors that influence the shear modulus (hence Rayleigh-wave dispersion) of a tissue is temperature, where the shear modulus often decreases with increasing temperature. In this talk, we discuss the temperature dependence of dispersion characteristics of Rayleigh waves propagating along the surface of soft tissues. Rayleigh-wave spectroscopy was performed at temperatures ranging from 25 to 40 °C using soft tissue phantoms that closely mimicked the physical properties human dermis. It was found that Rayleigh-wave dispersion exhibits high sensitivity to change in temperature, which could be used in real-time, noninvasive monitoring of skin temperature.

4:00

4pBAa11. Quantitative assessment of surface wave in kidney using optical coherence elastography. Hsiao-Chuan Liu (Mayo Clinic, 200 First St. SW, Rochester, MN 55905, liu.hsiao-chuan@mayo.edu), Piotr Kijanka, and Matthew W. Urban

Patients with chronic kidney disease (CKD) and renal transplant rejection are at high risk to develop advanced pathology including glomerular sclerosis and renal interstitial fibrosis. The elasticity of kidney tissues is associated with the disease progression and can be evaluated with ultrasound elastography. Understanding how tissue pathology changes the renal mechanical properties is crucial. Optical coherence elastography (OCE) might be a promising tool for evaluating renal tissues due to higher resolution. A phantom with 8% gelatin and 2% cellulose (50 mm) was fabricated to validate the ability of OCE to bulk materials and the results were confirmed by comparing the dataset obtained with an ultrasound-based Verasonics system as the ground truth. Excitation signals, composed with 4 cycles of a 400 Hz harmonic wave repeated at a rate of 38 Hz, were directly applied to the surface area with two directions and cortical regions of *ex vivo* porcine kidneys that had been cut open. Preliminary results showed that the velocity of surface wave on the cortical regions was near 1 m/s and corresponded with the result of shear waves from Verasonics system (1.2 m/s). Our study demonstrated that OCE has an ability to assess the mechanical properties of bulk materials and is a promising tool to evaluate mechanical properties of renal tissue with high resolution and sensitivity.

4:15

4pBAa12. Evaluating robustness of local phase velocity imaging in elastic and viscoelastic phantoms. Benjamin Wood (Radiology, Mayo Clinic, 1523 21st Ave. NE, Rochester, MN 55906, wood.benjamin@mayo.edu), Piotr Kijanka, and Matthew W. Urban (Radiology, Mayo Clinic, Rochester, MN)

Local Phase Velocity Imaging (LPVI) is a novel method of shear wave elastography (SWE) analysis, which provides phase velocity images at multiple frequencies. The primary components of this method include using two acoustic radiation force (ARF) pushes near the edges of the field-of-view. Fourier analysis of the spatiotemporal data is conducted to obtain maps of the phase velocity at multiple frequencies. A robustness study was conducted to evaluate the effectiveness of LPVI in elastic and viscoelastic phantoms with different mechanical properties, ARF focal depths and F-numbers, and shear wave frequencies using a linear array transducer. The results showed effective phase velocity reconstructions for most scanning parameters in the phantoms evaluated. We evaluated mean and standard deviations of phase velocities in the different phantoms for regions-of-interest defined in the images. Issues were seen with shear wave signal being distorted from the incident push when using low F-numbers as well as issues with attenuating shear waves with higher F-numbers and deeper focal depths. Future directions of this work will be to evaluate the robustness of LPVI in inclusion phantoms as well as testing on a curved array transducer.

Session 4pBAb

Biomedical Acoustics: Cavitation Bioeffects II

Hong Chen, Cochair
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Julianna Simon, Cochair
Graduate Program in Acoustics, The Pennsylvania State University, 201E Applied Science Building, University Park, Pennsylvania 16802

Invited Paper

1:30

4pBAb1. Modifying whole blood oxygen levels via acoustic droplet vaporization nucleated with an intravascular ultrasound system. Newsha Jahanpanah (I), Rachel P. Benton, Bin Bo (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH), Abigail R. Clark (Medical Sci. Baccalaureate Program, Univ. of Cincinnati, Cincinnati, OH), Karla P. Mercado-Shekhar (Biological Eng., Indian Inst. of Technol. Gandhinagar, Gandhinagar, Gujarat, India), and Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209, kevin.haworth@uc.edu)

Tunable oxygen scavenging via acoustic droplet vaporization (ADV) has been demonstrated in buffers. Whole blood presents a unique challenge due to the potent oxygen carrying capacity of hemoglobin. The objective of this study was to determine whether whole blood hemoglobin oxygen saturation (SO_2) and oxygen partial pressure (PO_2) could be modified via ADV. Perfluoropentane droplets were manufactured using high-speed shaking and microfluidic protocols. Droplets and Definity® were diluted in anticoagulated bovine whole blood at concentrations up to 2.5×10^{-3} ml/ml and 1.5×10^{-4} ml/ml, respectively. The whole blood ($SO_2 = 96\%$ and $PO_2 = 105$ mmHg) was pumped through an *in vitro* flow phantom at 37 °C and exposed to ultrasound (2.35 MHz, 1.5 MPa peak negative pressure, 40 or 120-cycles, 2 ms pulse repetition period) using an EkoSonic® catheter driven with a function generator and amplifier. SO_2 and PO_2 were measured using inline optical sensors. The occurrence of ADV was concurrent with decreases in SO_2 and PO_2. Increasing droplet concentrations and ultrasound pulse durations resulted in larger decreases in SO_2 and PO_2 with reductions to venous gas values ($SO_2 = 84\%$ and $PO_2 = 34$ mmHg). These results demonstrate a change in whole blood oxygen content can occur by ADV using an intravascular ultrasound catheter.

Contributed Paper

1:50

4pBAb2. Perfluorobutane phase shift nanoemulsion enables cavitation-mediated nonthermal ablation of brain tumors in rat. Chenguang Peng (Biomedical Eng., Boston Univ., 44 Cummington Mall B01, Boston, MA 02115, pcg11@bu.edu), Chanikarn Power, Tao Sun, Natalia Vykhodtseva, Yongzhi Zhang (Radiology, Brigham and Women's Hospital, Boston, MA), Tyrone M. Porter (Biomedical Eng., Boston Univ., Boston, MA), and Nathan McDannold (Radiology, Brigham and Women's Hospital, Boston, MA)

Ultrasound nonthermal ablation is an emerging noninvasive technology for transcranial ablation. Nonthermal ablation has been achieved previously through inertial cavitation of ultrasound contrast agents (UCA). However, it is possible to generate unwanted damage in pre-focal regions due to the presence of UCA throughout the beam path. We propose to use perfluorobutane phase shift nanoemulsions (PFB-PSNE) instead of MBs to limit cavitation-mediated damage to the focal region. PFB-PSNE consist of a lipid shell and a liquid perfluorobutane core which can be vaporized and nucleate inertial cavitation using ultrasound. In this study, we demonstrated PFB-PSNE facilitated ablation in rat brain tumor. Four Fischer rats were implanted with F98 glioma in the caudate putamen. Two rats were sonicated with PFB-PSNE, and the other two served as controls. The tumors were sonicated 8 days after tumor implantation, and the animals were sacrificed at day 13 to harvest the brains. The tumor growth was evaluated with magnetic resonance imaging and histology. Localized microhemorrhage and ischemia was observed in the targeted tumors. We did not observe effects in the beam path. In summary, this work demonstrates the feasibility of using activated PFB-PSNE to produce localized cavitation sources that result in controlled damage in a glioma model.

Invited Paper

2:05

4pBAb3. Microbubbles on kidney stones contribute to the twinkling artifact in humans. Julianna C. Simon (Graduate Program in Acoust., The Penn State Univ., University Park, PA and Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Penn State, 201E Appl. Sci. Bldg., University Park, PA 16802, jcsimon@psu.edu), Scott A. Zinck (Graduate Program in Acoust., Penn State Univ., University Park, PA), Eric Rokni (Graduate Program in Acoust., Penn State Univ., State College, PA), Jeffrey Thiel, Barbrina Dunmire, Bryan W. Cunitz (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), James R. Holm (Ctr. for Hyperbaric Medicine, Virginia Mason Medical Ctr., Seattle, WA), and Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

The color Doppler ultrasound twinkling artifact, a rapid color change that highlights kidney stones, was recently attributed to surface crevice microbubbles on *ex vivo* stones because twinkling was affected by changes in hydrostatic pressure. However, it was unclear whether crevice bubbles existed on *in situ* human kidney stones, and, if so, how smooth-surfaced stones could harbor the crevice bubbles that give rise to twinkling. Here, 8 human subjects with known kidney stones were exposed to 4 atmospheres absolute (ATA) while breathing air inside a hyperbaric chamber; twinkling was monitored and quantified with a research ultrasound system. At 3 and 4 ATA, twinkling was significantly reduced by an average of 35% and 39%, respectively ($p = 0.04$). Then, *ex vivo* kidney stones that twinkled were exposed to micro-computed tomography (μCT) and hypobaric pressures while submerged in water. Regions of low x-ray attenuation within the stone (i.e., microcracks) at 1 ATA expanded when the pressure was reduced to 0.1 ATA. These results support the theory that microbubbles are present on kidney stones in the human body and that microbubbles may be internal as well as external to the kidney stone. [Work supported by NSBRI through NASA NCC 9-58 and NIH DK043881.]

Contributed Paper

2:25

4pBAb4. High-speed microphotography of stone breakage with Holmium:YAG lasers. Yuri A. Pishchalnikov (Applaud Medical, Inc., 953 Indiana St., San Francisco, CA 94107, yurapish@gmail.com), William Behnke-Parks, Daniel Laser (Applaud Medical, Inc., San Francisco, CA), and Marshall Stoller (Dept. of Urology, Univ. of California, San Francisco, CA)

To better understand physical mechanisms of stone comminution in Holmium:YAG laser lithotripsy, high-speed microphotography was used to study the breakage of dry stones in air and hydrated stones in water. Surgically retrieved urinary calculi and hydroxyapatite pellets were pulsed with a Holmium:YAG laser at 0.2–1 J and recorded at ~250 000 fps using a high-speed camera (Shimadzu HPV-X2) mounted to a Nikon microscope. In air, direct light–stone interactions were seen to produce photothermal melting, vaporization, and micro-explosions. The most apparent damage to stones was due to the micro-explosions, which erupt the stone surface producing craters and pits. The micro-explosions, however, were observed to progressively diminish with number of laser shots, typically stalling after several shots. Increasing the laser energy or moving the fiber tip to a new position was seen to restore the occurrence of micro-explosions before stalling a second time. In water, the laser produced cavitation bubbles, the collapse of which continued to fragment, pit, and erode the stone after each laser pulse. This study suggests that cavitation bubbles contribute strongly to stone comminution. Cavitation bubbles may have a particularly important role in sustaining micro-explosions after initial laser shots.

Session 4pCA

Computational Acoustics, Physical Acoustics, Underwater Acoustics, Acoustical Oceanography, Signal Processing in Acoustics, and Structural Acoustics and Vibration: Parabolic Equation Methods Across Acoustics

Jennifer Cooper, Cochair
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Michelle E. Swearingen, Cochair
US Army ERDC, Construction Engineering Research Laborartory, P.O. Box 9005, Champaign, Illinois 61826

Subha Maruvada, Cochair
U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, Maryland 20993

Chair's Introduction—1:15

Invited Papers

1:20

4pCA1. Modeling 3-dimensional sound propagation using the parabolic equation. Kevin D. Heaney (Appl. Ocean Sci., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com) and Richard L. Campbell (OASIS, Inc., Lexington, MA)

For low frequency sound (below 50 Hz), all ocean propagation is effectively shallow water, as the impact of the seafloor topography impacts the direction and level of acoustic propagation. In this paper, we present a parabolic equation (PE) model, based upon the Range-dependent Acoustic Model (RAM), developed by Michael Collins, which was extended to handle parts of this 3-dimensional acoustic propagation problem. The model, which applies the Pade propagation operator in cross range, after each range step, is well suited to global scale problems where the effects of diffraction and refraction dominate. A quantitative comparison of observations of undersea explosions detonated off the coast of Japan measured from Chile will be presented. In addition to the presentation of this model and it's application to several observed 3-D propagation effects, a collection of hybrid models used historically to address the out-of-plane propagation will be presented.

1:40

4pCA2. Use of the parabolic equation method for local infrasound propagation predictions. Michelle E. Swearingen (US Army ERDC, Construction Eng. Res. Laborartory, P.O. Box 9005, Champaign, IL 61826, michelle.e.swearingen@usace.army.mil) and Ross E. Alter (US Army ERDC, Hanover, NH)

Local infrasound (<150 km) propagation is strongly influenced by range- and time-dependent meteorological conditions. The Parabolic Equation computational method provides an efficient and flexible tool for predicting infrasound propagation in this regime. Over these longer distances, the influence of spatial and temporal variations in the meteorological conditions can be significant. Meteorological conditions extracted from simulations using the Weather Research and Forecasting (WRF) model are incorporated by updating the effective sound speed profile as a function of propagation range. By examining horizontal grid increments between 1 and 15 km and temporal resolutions between 4 and 60 seconds, we begin to quantify the potential variability in predictions at ranges of up to 150 km, for frequencies between 1 and 20 Hz. This procedure will inform a broader discussion of the application of the PE method to infrasound prediction. [Work funded by the Assistant Secretary of the Army (Acquisition, Logistics, and Technology) [ASA(ALT)] under 62784/T40. Distribution Statement A: Approved for public release; Distribution is unlimited.]

2:00

4pCA3. Extra-wide-angle parabolic equation for wave propagation in inhomogeneous media. Vladimir E. Ostashev (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd, Hanover, NH 03755, vladimir.ostashev@colorado.edu), D. Keith Wilson, Michael Muhlestein, Michael Shaw (U.S. Army Engineer Res. and Development Ctr., Hanover, NH), Michelle E. Swearingen (U.S. Army Engineer Res. and Development Ctr., Champaign, IL), and Sarah McComas (U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS)

To describe wave propagation at large angles with respect to a nominal direction, wide-angle parabolic equations (WAPEs) have been widely used in atmospheric and ocean acoustics, geophysics, electromagnetic wave propagation, and other fields of physics. This paper considers an application of an extra-wide-angle parabolic equation (EWAPE) for such problems. For small variations of the refractive index of a medium, the EWAPE describes wave propagation up to 90 deg with respect to the nominal direction and is more general than the WAPEs used in the literature. The EWAPE can be written in an integral form or as a pseudo-differential equation and may be solved by a split-step spectral algorithm or the Padé series expansions of the pseudo-differential operators. The EWAPE is also generalized to large variations in the refractive index, sound propagation above an impedance boundary, and a moving medium with small or large Mach numbers. For sound propagation in a moving medium, the EWAPE enables derivation of new WAPEs which are accurate and simpler to implement than those currently available in the literature. Numerical examples illustrating the application of these new WAPEs to sound propagation in a moving atmosphere are presented.

Contributed Papers

2:20

4pCA4. Model-data comparison of the parabolic equation approximation for long-range propagation, time spreading, and channel bandwidth estimation. Paul C. Hines (Dept. of Elec. and Comput. Eng., Dalhousie Univ., PO Box 15000, Halifax, NS B3H 4R2, Canada, phines50@gmail.com), Terry Deveau (JASCO Appl. Sci., Dartmouth, NS, Canada), Elizabeth Kusel (JASCO Appl. Sci., Portland, OR), David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., Halifax, NS, Canada), Dainis Nams, and Matt Coffin (GeoSpectrum Technologies, Inc., Dartmouth, NS, Canada)

Underwater communications and sonar measurements in the Arctic are challenging due to the remoteness and the vast area to be covered. Acoustic modeling is critical to support measurements since it can help inform experimental plans and extend the usefulness of results to other areas. Low-frequency systems are generally required to deal with the vast ranges being covered and to mitigate the high-frequency signal loss caused by under-ice roughness. The Parabolic Equation Approximation (PE) is well suited to the low-frequency band but properly accounting for the rough, elastic, ice layer is challenging. A recent project has integrated a rough, elastic, ice layer into a long-range, time-domain PE model to provide complementary modeling support for a newly developed low-frequency Arctic sonar system. This presentation will review propagation measurements and the supporting PE modelling from a deep water experiment off the Scotian Shelf at frequencies from 16 to 250 Hz and ranges up to 700 km, prior to deployment in the Arctic.

2:35

4pCA5. Modeling nonlinear fractional diffusion-wave equations with Burgers-type evolution equations. Blake E. Simon (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, blakesimon8@utexas.edu) and Mark F. Hamilton (Appl. Res. Labs. and Walker Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

Fractional partial differential equations (FPDEs) with a time derivative of fractional order are used to describe wave motion in complex viscoelastic media with non-traditional equations of motion. Kappler *et al.* [*Phys. Rev. Fluids* **2**, 114804 (2017)] derived a fractional diffusion-wave equation for a nonlinear Lucassen wave propagating along an elastic layer coupled to a viscous substrate. The fractional time derivative of order 3/2 in the linear form of this equation lies midway between order 1 for a diffusion process described by a parabolic equation, and order 2 for the traditional hyperbolic wave equation. The inclusion of nonlinear elasticity tends to inhibit purely progressive wave motion that is associated with classical nonlinear plane waves in fluids and solids, and which is described accurately by parabolic-like, Burgers-type evolution equations. In this work, a general FPDE is analyzed in a parameter space consisting of varying nonlinearity and time fractional orders. The focus is on conditions under which the FPDE can be modeled accurately with a Burgers-type evolution equation for progressive wave motion. The method of lines in combination with a general Runge-Kutta method for forward integration is used for numerical analysis. [B.E.S. is supported by the ARL:UT McKinney Fellowship in Acoustics.]

2:50–3:05 Break

Invited Paper

3:05

4pCA6. The parabolic approximation in therapeutic ultrasound. Joshua Soneson (Appl. Mech., FDA, 10903 New Hampshire Ave., Silver Spring, MD 20993, joshua.soneson@fda.hhs.gov)

An ideal mathematical model is both accurate and tractable. It must predict the physics of interest and produce solutions in an efficient manner. In the field of therapeutic ultrasound, where simulations of finite-amplitude sound beams are an important adjunct to bench testing, the parabolic approximation of the wave equation often strikes a balance between these two requirements. In this presentation the common approaches to computing ultrasound fields are introduced and a family of high-order parabolic models is derived and analyzed. This analysis provides the framework for addressing a challenge often encountered in medical ultrasound modeling: controlling spurious effects associated with steep gradients in the source boundary condition. The accuracy of the resulting high-order propagation models is assessed by comparing against solutions of the full wave equation and the degree to which these models capture the salient characteristics of ultrasound beams is analyzed. This approach brings to therapeutic ultrasound a “wide-angle” parabolic model with improved diffraction modeling capability and no penalty to computational efficiency.

Contributed Paper

3:25

4pCA7. Reflection and refraction of sound waves on a smooth and the non-flat boundary between two liquids. Nikolai Maltsev (none, 1467 Leaftree cir, San Jose, CA 95131, admin@asymptotus.com)

The problem of the scattering of sound waves by the boundary between two liquids has an elementary solution in a case of flat boundary and can be reduced to a solution of an integral equation with the singular oscillating kernel. In practical applications, wast majority of boundaries are treated as random surfaces and approximate methods produce only some statistical moments of the sound field. When the boundary is smooth and non-flat there is no simple method which can produce physically consistent results. This work presents an approach, which exploits a smoothness of the boundary to build an orthogonal coordinate system and rewrite wave equations at both sides of the surface in this system. For a given incident field of sound pressure and particles velocities by satisfying the boundary conditions and applying the split-step method and FFT we build a reflected and refracted field of sound pressure and particles velocity in both media, producing physically reliable interference patterns.

Invited Paper

3:40

4pCA8. Application of elastic parabolic equation solutions to acoustic reverberation in ice-covered underwater environments. Scott D. Frank (Dept. of Mathematics, Marist College, 3399 North Ave., Poughkeepsie, NY 12601, scott.frank@marist.edu) and Anatoliy N. Ivakin (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Parabolic equation solutions can be obtained for underwater acoustic environments where elastic boundaries affect the acoustic propagation. One example of this situation is in the Arctic where a (potentially rough) ice layer is on top of the water column. The parabolic equation method shown here rigorously applies the zero-traction and fluid-elastic interface boundary conditions to obtain full field Green's function estimates near the ice-water interface. This solution is crucial for calculation of recently derived reverberation estimates that require both horizontal and vertical derivatives of the complex acoustic field. Monostatic reverberation estimates for a rough under-ice interface are calculated for an upward refracting water sound speed profile and an elastic ocean bottom with a nearly fluid mud layer. These calculations could be used, for example, to estimate areal distribution of the ice layer roughness from azimuth-time dependence of acoustic reverberation measured in Arctic environments. Performance of the Padé rational function approach in the presence of thin ice cover and low shear speed ocean bottom layers will be addressed.

Contributed Paper

4:00

4pCA9. Application of a multi-sector parabolic equation approach to compute acoustic scattering by non-canonically shaped impenetrable objects. Adith Ramamurti (US Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, adith.ramamurti@nrl.navy.mil) and David C. Calvo (US Naval Res. Lab., Washington, DC)

While parabolic equation (PE) methods have seen great use in ocean acoustic waveguide propagation, it is not as widely known that a multi-sector PE method, originally proposed by Levy and Zaporozhets [*J. Acoust. Soc. Am.* **103**, 735–741 (1998)], can be applied to compute the scattered field about an object. Previous work was only able to benchmark the method with canonical geometries. In this talk, we demonstrate favorable agreement between free-field PE-based and finite-element based computations, the latter being taken as a benchmark, in two and three dimensions for non-canonically shaped impenetrable objects. Run-time scaling comparisons are also presented, which demonstrate the great advantage of the PE method in the high-frequency limit when a target is several or more wavelengths long. We also show how wide-angle PE and multiple-scattering corrections can be incorporated to treat concave scatterers. [Work sponsored by the Office of Naval Research.]

Session 4pEDa

Education in Acoustics and Musical Acoustics: Selecting a Textbook for Teaching an Acoustics Course

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg, University Park, Pennsylvania 16802

Jack Dostal, Cochair
Physics, Wake Forest University, P.O. Box 7507, Winston-Salem, North Carolina 27109

Invited Papers

1:00

4pEDa1. In search of the elusive perfect textbook. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, dar119@psu.edu)

During my 24-year teaching career so far, besides teaching undergraduate service courses from standard textbooks, I have also taught a variety of advanced courses in acoustics, both at undergraduate and graduate levels. Some courses were well suited to a single textbook, especially when a well-written textbook existed matching course topics. Other times, it was difficult to find a single book covering all course topics at an appropriate level. Sometimes, the "best fit" book was no longer in print, and available alternatives were unsatisfactory for various reasons. For a few courses, I changed books every semester several times before finally finding something that worked. In this talk, I will discuss my experiences selecting, evaluating, and adopting textbooks for courses which I either developed myself (physics of waves, noise control engineering, acoustics of musical instruments) or for which I inherited an established syllabus from another faculty member (fundamentals of acoustics, theoretical acoustics, classical mechanics). I will discuss selection criteria I used to evaluate textbooks, including topical coverage, difficulty level, readability, homework problems, reviews, reputation, and price. I will also discuss what I did when I could not find a satisfactory textbook.

1:15

4pEDa2. Selecting a new textbook for a graduate level course on vibration and fluid acoustics. Brian E. Anderson (Dept. of Phys. & Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, bea@byu.edu) and Scott D. Sommerfeldt (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

At Brigham Young University we offer a graduate level course on the fundamentals of acoustics. This course has traditionally covered the fundamentals of vibration in masses and springs, strings, bars, membranes and plates in the first half of the course. The second half of the course covers the derivation of the wave equation, wave propagation, reflection and transmission phenomena, and source radiation problems. For a couple decades, we used Kinsler and Frey's "Fundamentals of Acoustics" book but the rising costs, lack of updates, and dissatisfaction with the rigor of the book motivated us to find a new text. For a few years, we used Kinsler and Frey for the first half of the course (vibrations) and also Blackstock's "Fundamentals of Physical Acoustics" for the second half (fluids). Two books meant even higher costs, although the Kinsler and Frey text was also used in another class at that time. When we switched instructors for the course, Anderson was asked to review Garrett's "Understanding Acoustics" text and he ended up adopting it for this course. This talk will review our experiences in switching textbooks and discuss our reasons for selecting our current text and how we are using it in class in conjunction with pre-class reading quizzes, which has helped increase classroom discussions.

1:30

4pEDa3. Choosing a textbook for the way your students learn. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, andy.piacsek@cwu.edu)

This presentation describes the challenges, opportunities, and lessons learned in the selection and use of textbooks for a 10-week course in acoustics for undergraduate physics majors at the junior or senior level at Central Washington University. This course emphasizes both theoretical and experimental skills; topics include damped harmonic systems, vibration in strings, bars, and membranes, plane and spherical wave propagation in air, and basic geometric acoustics. Two textbooks are employed for this course: the recently published "Understanding Acoustics" by Steve Garrett and the fourth edition of "Fundamentals of Acoustics" by Kinsler and Frey, revised and extended by Coppins and Sanders. Although both cover essentially the same topics, they have different attributes that facilitate learning at the undergraduate level. They are supplemented by online materials, interactive simulations, and journal articles. Examples of how these texts are used, and assessments of their efficacy, will be discussed.

1:45

4pEDa4. Comparing three textbook choices for an introductory physics of music class. Jack Dostal (Dept. of Phys., Wake Forest Univ., P.O. Box 7507, Winston-Salem, NC 27109, dostalja@wfu.edu)

Wake Forest's Physics of Music is an introductory science class focused on musical acoustics. It is commonly taken by music students or non-science majors seeking a science credit. The course has been taught using three different texts (Rossing *et. al*., Heller, and Berg and Stork) in the past few years. The texts vary in approach, focus, and breadth. I will describe some successes and difficulties I have experienced while teaching from these texts. I will also comment on additional texts such as Hall and Campbell & Greated which I regularly use to supplement the others.

2:00

4pEDa5. Investigating the connection between physics and music in an introductory acoustics course. Jill A. Linz (Phys., Skidmore College, 815 N. Broadway, Saratoga Springs, NY 12866, jlinz@skidmore.edu)

Sound and music is an introductory acoustics course offered at Skidmore College in Saratoga Springs, NY. The target audience is primarily non-science majors fulfilling their math and science requirements. Over the span of its 25 + year history it has regularly attracted students interested in majoring in music technology and acoustics. Emphasis of the course is placed on the physics of music and music theory with attention paid to the interconnection between atomic physics and music. Fourier's Theorem is used in making this connection. Sound editing programs such as Amadeus allow students to visualize recorded or synthesized sounds, and then analyze their spectra using Fourier analysis. Most course activities have been developed in accordance with Physics Education Research techniques and are designed using a discovery-based approach. Several activities from Sound & Music were recently highlighted in an article published online by Physics Today titled *Composing Atom Music*. Although Rossing, Moore & Wheeler's *The Science of Sound* is the primary text, course materials are drawn from numerous sources. This presentation will focus on the course design, how the text book is used and will discuss the design and implementation of sample class activities.

2:15

4pEDa6. Using acoustics to prepare undergraduate physics majors for the fundamentals of quantum physics. Kurt Hoffman (Phys., Whitman College, 345 Boyer Ave., Hall of Sci., Whitman College, Walla Walla, WA 99362-2067, hoffman@whitman.edu)

As part of curricular changes in our Physics curriculum, I developed an acoustics lecture course as a bridge between General Physics and our modern physics course. The goals of this course were to provide students with new mathematical tools (solving the wave equation, using complex numbers, boundary conditions) while also making physical connections between the mathematics and a system they could work with in a classroom. My aim was to prepare them for the mathematics to solve quantum problems so that they could spend more time wrestling with abstract quantum concepts. I used a textbook for vibrations and waves as a basis for the course and supplemented the text with acoustics specific content. I will present an outline of my approach and feedback from students regarding the utility of using acoustics as a primer for the more abstract content encountered in a Modern Physics course. Surveys and student reflections solicited after completing both courses will be discussed. My presentation will focus on whether the benefits of using acoustics to help students prepare for quantum physics warrants inclusion in the standard curriculum.

2:30

4pEDa7. Teaching musical meter to school-age students through the Ski-Hill graph. Andrea Calilhanna (MARCS Inst. for Brain, Behaviour and Development, 2 Kayla Way, Cherrybrook, Sydney, New South Wales 2126, Australia, A.Calilhanna@westernsydney.edu.au)

This thesis, *Teaching Musical Meter to School-Age Students Through the Ski-Hill Graph*, aims to demonstrate the "pedagogability" of modern meter theory, that is, that new scholarship on meter can translate into a coherent and practically implementable instructional curriculum, with various advantages for school-age students. The curriculum model developed in the thesis is derived from Richard Cohn's work on and approaches to meter theory, the first comprehensive theory and approach to teaching and learning meter which focuses on the embodied psychoacoustic experience of sound and graphic representations of meter through mathematical music theory rather than notation-based understandings. The materials set out in the thesis demonstrate ways students might be taught to articulate their experience of meter. A unified approach, it incorporates Cohn's ski-hill graph and other instruments of mathematical music theory such as the SkiHill app (Milne, 2018) and XronoBeat app (Milne, 2019) numbering for counting meter, cyclic graphs, and beat-class theory. The thesis demonstrates how traditional Western music theory has evolved due largely to non-Western influences to now possess the capacity for analyzing music from around the world. In addition, the outcomes it anticipates is a deeper engagement with music in classroom settings.

Session 4pEDb

Education in Acoustics: Take 5's

Jack Dostal, Chair
Physics, Wake Forest University, P.O. Box 7507, Winston-Salem, North Carolina 27109

You are invited to bring your favorite acoustics teaching ideas. Choose from the following: short demonstrations, teaching devices, or videos. The intent is to share teaching ideas with your colleagues. If possible, bring a brief, descriptive handout with enough copies for distribution. Spontaneous inspirations are also welcome.

Sign up at the door for a five-minute slot before the session begins. If you have more than one demo, sign-up for two consecutive slots.

THURSDAY AFTERNOON, 5 DECEMBER 2019 CONTINENTAL/CRYSTAL, 1:00 P.M. TO 5:25 P.M.

Session 4pNS

Noise, Physical Acoustics, and Computational Acoustics: Supersonic Jet Aeroacoustics II

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

Alan T. Wall, Cochair
Battlespace Acoustics Branch, Air Force Research Laboratory, Bldg. 441, Wright-Patterson AFB, Ohio 45324

Chair's Introduction—1:00

Invited Papers

1:05

4pNS1. Comparative studies on lift-off acoustic environments among flight data of the H-IIB launch vehicle, sub-scale acoustic tests and numerical simulations. Ayano Mori (JAXA, Mazu, Kukinaga, Minamitane-cho, Kumage-gun, Kagoshima 891-3793, Japan, mori.ayano@jaxa.jp), Wataru Sarae (JAXA, Ibaraki, Tsukuba, Japan), Keita Terashima (JAXA, Tsukuba, Japan), Seiji Tsutsumi (JAXA, Sagamihara, Kanagawa, Japan), Masao Takegoshi (JAXA, Kakuda, Japan), and Hiroaki Kobayashi (JAXA, Kanagawa, Japan)

A 1/42-scale acoustic tests were conducted to predict the lift-off acoustic environments of the H3 launch vehicle currently under development in Japan. The H3 acoustic level is estimated by relative comparison among sub-scale acoustic tests of the H3 / the H-IIB and flight data of the H-IIB. In this presentation, comparative studies between sub-scale tests and full-scale flight data for the H-IIB have been conducted as well as overall sound pressure level. CFD numerical simulations were also conducted to comprehend the phenomena. These provide further confidence that the sub-scale acoustic tests can favorably be used to verify the H3 lift-off acoustic environments.

1:25

4pNS2. Visualization and acoustic-triggered conditional sampling analysis of acoustic phenomena with screech-tone of an over-expanded supersonic jet. Miho Takemura (The Univ. of Tokyo, 5-1-5 Kashiwanoha, Kashiwa, Chiba 277-8561, Japan, 6859921335@edu.k.u-tokyo.ac.jp), Koji Okamoto (The Univ. of Tokyo, Kashiwa, Chiba, Japan), and Susumu Teramoto (The Univ. of Tokyo, Bunkyo, Tokyo, Japan)

In the liftoff of launch vehicles and supersonic aircrafts, jet noise is required to be reduced because it causes various problems. To reduce the influence of jet noise, it is significant to clarify the source location and the direction of propagation as well as the spectrum of each acoustic phenomenon. For this purpose, the authors have proposed the acoustic-triggered conditional sampling analysis to obtain acoustic information from high-speed schlieren movies. In the authors' previous studies, this method was applied to correctly and under expanded jets, and the broadband phenomena were clarified, such as Mach wave radiation and shock associated noise. In this study, this method was applied to an over-expanded jet, in which screech tone is observed in addition to the broadband acoustic phenomena. First, it was found that the broadband phenomena and screech tone could be observed separately by this method with choosing the microphone location for trigger detection. Then, two screech tone noises of different frequencies are observed, and it was clarified that they are propagating to different directions from different source locations. Also, the acoustic phenomenon generated by impingement of the same over-expanded jet on an inclined flat plate will be discussed.

1:45

4pNS3. Flow and acoustic investigations into the impingement of single and multiple jets onto a jet blast deflector. Karthikeyan Natarajan (Experimental AeroDynam. Div., CSIR-National Aerosp. Labs., EAD, PB 1779, Old Airport Rd., Bangalore 560017, India, nkarthikeyan@nal.res.in) and Lakshmi Venkatakrishnan (Experimental AeroDynam. Div., CSIR-National Aerosp. Labs., Bangalore, Karnataka, India)

Current space launch vehicles, employ a combination of Solid Rocket Boosters and Liquid Rocket Engines to provide the thrust required during the first stage of lift-off. Such combinations lead to the deployment of a multitude of jets, whose interactions with the Jet Blast Deflector(JBD) can have a significant bearing on the overall acoustic loading experienced by the vehicle. Flow and acoustic fields of multiple jets impinging on a jet blast deflector has not been adequately explored in the past. The present work investigates the flow and acoustic characteristics of four identical jets impinging on a generic JBD and compares them against the results for a single equivalent jet impingement. All the jets studied, issue from conical C-D nozzles with design Mach number of 2.0. The combined mass flow from the multiple jets was equal to that of the single jet. The flow field characterization was carried out using schlieren and the acoustic field was studied using microphone measurements inside an anechoic chamber. Two different arrangements of the multiple jets were studied to understand their interactions with the JBD and their influence on the flow and acoustic field generated.

Contributed Papers

2:05

4pNS4. Comparison of exhaust duct designs in launch pads for reduction of rocket jet noise. Taeyoung Park (School of Mech. Eng., Yonsei Univ., Eng. Bldg. A391, 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, pty0948@yonsei.ac.kr), Inman Jang, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., Seoul, South Korea), H. S. Joo, Seunghoon Kang (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 Jeongwon Park (Agency for Defense Development, Daejeon, South Korea)

An exhaust duct is commonly installed in a launch pad to reduce the intense noise generated by the rocket exhaust jet at liftoff. In this paper, we investigate the effects of varying the length and shape of the exhaust duct by comparing the noise reduction performance of several different designs. The sub-scale exhaust ducts tested are variations of the original design by Tsutsumi *et al.* [AIAA Paper 2015-1007], colloquially known as "Seiji's dog house," and experiments are performed using cold jets of Mach 1.8 and 3.1, respectively. Linear and semi-circular microphone arrays are employed near the nozzle and at a radial distance of 100 nozzle diameters from the center of the duct outlet. A design guideline as to the length and the outlet shape of the exhaust duct, leading to the best noise reduction performance, is discussed. [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.]

2:20

4pNS5. Passive jet noise reduction using contoured inserts. Nathan E. Murray (National Ctr. for Physical Acoust., The Univ. of MS, 1 Coliseum Dr., University, MS 38677, nmurray@olemiss.edu)

Successful supersonic jet noise reduction requires an approach that simultaneously (1) breaks up coherent structures in the jet shear layer turbulence, (2) reduces convection velocity of the acoustic sources, and (3) weakens the shock structure in the plume. If any one of these is done in isolation the result is typically a shift in frequency content and not an overall reduction in source amplitude. Using a set of contoured inserts placed within the expansion section of the nozzle allows all three of these design goals to be addressed. The contour weakens the shock structure while simultaneously generating strong streamwise vorticity the enhances mixing in the jet shear layer. Assuming that future high-performance aircraft would possess a variable area ratio nozzle with independent exit and throat area controls, multi-objective design optimization has yielded a design that maximizes noise reduction with minimal thrust penalty in a variable area ratio framework. Model-scale acoustic data are presented to demonstrate the efficacy of the design.

2:35

4pNS6. Far-field acoustical measurements during GEM-63 static firings. Michael S. Bassett (Dept. Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, MichaelScottBassett@physics.byu.edu), Francisco J. Irarrazabal, Reese D. Rasband, Daniel J. Novakovich, Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Raiarii Jithame, and Jon P. Johnson (Phys., Brigham Young Univ. Idaho, Rexburg, ID)

This paper expands on prior measurements and analyses of noise data from static rocket firings in northern Utah [e.g., see B. O. Reichman *et al.*, *Proc. Mtgs. Acoust.* **25**, 045006 (2017)]. Far-field acoustical measurements were made on two separate occasions for the GEM-63 motor, a 1660 kN-thrust solid-fuel booster. Measurements were taken at angles between 40 and 120 deg (relative to the plume direction) and distances of 985–1832 diameters from the nozzle, using both 6.35 mm and 12.7 mm diameter microphones and at sampling rates of at least 50 kHz. This paper discusses the measurement setups, as well as other appreciable features relevant for data analysis, such as the local meteorology and surrounding terrain. Statistical, spectral, and correlation analyses are used to understand the frequency and temporal characteristics of the noise as a function of angle, as well as consistency between the two motor tests. [Work supported in part by NSF and NASA.]

2:50

4pNS7. Spectral analysis of directivity and source location of turbulent jet noise. Oliver T. Schmidt (Mech. and Aerosp., Univ. of California San Diego, 9500 Gilman Dr., MC 104-44, La Jolla, CA 92093, oschmidt@ucsd.edu) and Akhil Nekkanti (Mech. and Aerosp., Univ. of California San Diego, La Jolla, CA)

We use spectral proper orthogonal decomposition (SPOD) of two high-fidelity numerical simulation databases of turbulent jets at $M=0.9$ and $M=1.5$ to investigate the directivity of jet noise. By spatially windowing the SPOD, we investigate the dependence of the far-field pressure PSD as a function of radiation angle on frequency, and study the source location of different modal contributions. The low-rank behavior of the far-field sound is addressed by reconstructing the total PSD using the leading SPOD modes. To understand the energetically most efficient mechanisms of energy transfer from hydrodynamic fluctuations in the near-field to the acoustic far-field, we further solve a generalized SPOD eigenvalue problem that was originally devised for the empirical computation of resolvent modes from data. Here, we specialize this method to optimal input-output relations between different components of the state vector.

3:05

4pNS8. Partial field decomposition of the simulated noise from a highly heated laboratory-scale jet. Kevin M. Leete (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, KML@byu.edu), Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Junhui Liu (Naval Res. Lab., Washington, DC), and Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH)

The decomposition of jet noise fields into self-coherent and mutually incoherent parts is an important step preliminary to the application of acoustical holography techniques. Cross-spectral matrices of point arrays sampled from large eddy simulations of a laboratory-scale jet operating at a high temperature ratio are decomposed using a singular value decomposition to obtain partial fields. The shape and relative strengths of the individual partial fields highlight the difference between the coherent radiation in the aft direction and the incoherent portion of the radiation towards the sideline. Azimuthally, levels are mostly constant but require multiple partial fields to accurately represent the low coherence at high frequency.

3:20–3:35 Break

3:35

4pNS9. Connecting high-power jet noise characteristics with human annoyance: Physical characterization. Aaron Vaughn (Brigham Young Univ., C110 ESC, Provo, UT 84602, aaron.burton.vaughn@gmail.com), Kent L. Gee (Brigham Young Univ., Provo, UT), Micah Downing, Michael M. James, Matt Calton (Blue Ridge Res. and Consulting LCC, Asheville, NC), and Alan T. Wall (Battlespace Acoust. Branch, Air Force Res. Lab., Wright-Patterson AFB, OH)

Physical characterization of annoying components of jet noise will aid in future noise mitigation efforts. Crackle perception is a significant contributor to annoyance produced by supersonic jet noise and is related to the reception of acoustic shocks. The skewness of the pressure time derivative, or derivative skewness, is sensitive to the presence of acoustic shocks and has been shown through listening studies to be indicative of crackle perception. Extensive measurements of a tied-down high-performance military aircraft allow for the examination of spatial variation and source characterization of events related to crackle perception in the field. A variety of factors contribute to the derivative skewness in the field, such as source-receiver distance, apparent source origin, or whether a microphone is placed on or elevated off the ground. Spatially, the derivative skewness often peaks in the maximum radiation region, however, deviations exist at lower engine conditions with sudden increases in derivative skewness in the aft region. An event-based beamforming method is implemented to examine the source region, with resulting directivities lending insight into far-field crackle perception. Characterization of crackle perception using derivative skewness as an indicator will increase understanding of mechanisms contributing to jet noise annoyance. [Work Supported by AF SBIR Program.]

Invited Paper

3:50

4pNS10. Connecting high-power jet noise characteristics with human annoyance. Micah Downing, Michael M. James, Matt Calton, Alexandria R. Salton (Blue Ridge Res. and Consulting, 29 N Market St., Ste. 700, Asheville, NC 28801, Alex.Salton@blueridgeresearch.com), Kent L. Gee, Aaron Vaughn (Brigham Young Univ., Provo, UT), and Gregory H. Wakefield (Univ. of Michigan, Ann Arbor, MI)

DoD has funded the development of advanced modeling tools for community noise exposures and for exploring noise reduction techniques. However, inadequate understanding of high-power jet noise sources and the effects of radiated noise on people are both limiting factors in mitigating community noise exposure. In the meantime, both ground personnel and community noise exposure lead to annoyance and disturbance, costly noise mitigation, and burdensome restrictions on operations. Therefore, DoD is exploring acoustical metrics that may better correlate jet noise exposure to human response. More broadly, DoD and the jet noise research community desire an increased understanding of jet noise source characterization to best focus on promising strategies for reducing jet noise impacts. This research project seeks to address these two needs by increasing the understanding of tactical jet noise sources and by correlating their physical noise features, such as noise source distribution, directivity, and shock content, with human response related to annoyance, disturbance, and the perception of crackle. This presentation will provide an overview of the project's objectives. [Work Supported by AF SBIR Program.]

4:10

4pNS11. Connecting high-power jet noise characteristics with human annoyance: Listener trials. Matt Calton (Blue Ridge Res. and Consulting, 29 N. Market St., Ste. 700, Asheville, NC 28801, matt.calton@blueridgeresearch.com), Michael M. James, Micah Downing (Blue Ridge Res. and Consulting, Asheville, NC), Aaron Vaughn, Kent L. Gee (Brigham Young Univ., Provo, UT), and Gregory H. Wakefield (Univ. of Michigan, Ann Arbor, MI)

Improved high-fidelity representations of jet noise characteristics will provide better inputs for environmental noise models, thereby assisting in improved mitigation efforts for reducing community noise exposure around military airfields. A series of initial listener trials has been conducted to relate jet noise characteristics with human response. The objective of these trials is to capture the unique perceptual features in high-power jet noise and properly represent their induced psychoacoustic response. One promising metric is the skewness of the pressure time derivative since it correlates with the strength and density of shocks in a jet noise waveform. This additional metric may aid in evaluating community noise exposures by identifying the temporal and spectral jet noise characteristics associated with human response. This presentation will provide an overall review of the current findings. [Work Supported by AF SBIR Program.]

4:25

4pNS12. Furthering resolvent-based jet noise models. Ethan M. Pickering (Mech. Eng., California Inst. of Technol., 1200 East California Boulevard, Mail Code MC 104-44, Pasadena, CA 91125, pickering@caltech.edu) and Tim Colonius (Mech. Eng., California Inst. of Technol., Pasadena, CA)

Resolvent analysis continues to provide promising results for modeling the hydrodynamic near-field and acoustic far-field in turbulent jets, particularly when compared to modes deduced through spectral proper orthogonal decomposition (SPOD) of high-fidelity large eddy simulations (LES). Although previous studies have shown that the acoustic field for supersonic jets is of low-rank and primarily described by the Kelvin-Helmholtz instability, the agreement between the acoustically optimal resolvent mode (Kelvin-Helmholtz) and optimal SPOD mode is still lacking. The discrepancy is due to the spatial coloring of turbulent mechanisms in supersonic jets and presents a challenge for reconstructing the acoustic field, as well as the full field flow statistics. To account for coloring, additional (i.e., suboptimal) resolvent modes, associated with the Orr mechanism, must be included and their correlation to other modes determined. Here, we estimate the coloring between resolvent modes by projecting onto an ensemble of LES realizations and reconstructing the realizations in the resolvent basis. The associated projection coefficients provide an ensemble of observations which inherently possess the statistical information necessary to reconstruct the flow and to which we propose a reduced-order stochastic model. We find that the inclusion of a few resolvent suboptimal modes (i.e., Orr-type) allows for modeling of the acoustic field to within 2 dB and increases the region of acoustic agreement when compared to a single mechanism model.

4:40

4pNS13. Spectral comparisons between ground runup and flyover operations of a high-performance military aircraft. Jacob A. Ward (Phys., Brigham Young Univ., N243 ESC, Provo, UT 84602, jacob.ward@live.com), Kent L. Gee (Phys., Brigham Young Univ., Provo, UT), and Micah Downing (Blue Ridge Res. and Consulting, Asheville, NC)

Prior work has shown differences between ground runup and flyover noise levels for military aircraft [B. O. Reichman *et al*., AIAA Paper 2018-3614]. This presentation describes a continuation of that effort to better understand spectral differences as a function of propagation distance, angle, and engine power. Acoustical measurements of a high-performance military jet aircraft made during both ground runup and flyover operations are described. The ground runup and flyover spectra show substantial differences in both shape and levels. In particular, more high-frequency energy is seen in the forward direction in the flyover data, which is believed to be from nonlinear propagation. This analysis examines in greater detail the increased nonlinearity in the forward direction, which is important to improved modeling of the noise from these aircraft. [Work supported by AFCEC.]

4:55

4pNS14. Comparative analysis of three falcon 9 launches. Logan T. Mathews (Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, lmathew3@byu.edu), Reese D. Rasband, Daniel J. Novakovich, Francisco J. Irarrazabal, Aaron Vaughn (Phys. and Astronomy, Brigham Young Univ., Provo, UT), Pauline Nelson (Phys., Brigham Young Univ. - Idaho, Rexburg, ID), and Kent L. Gee (Phys. and Astronomy, Brigham Young Univ., Provo, UT)

This study investigates the noise from three separate Falcon 9 vehicle launches from Vandenberg Air Force Base, as measured within the community of Lompoc, CA. Although one focus of these tests was to test a variety of acoustical and weather instrumentation systems and configurations, the launches also provides an opportunity to study and compare the launch noise measured under potentially different weather conditions. An analysis of the levels and spectra is shown for different time periods. The prevalence of acoustic shocks is examined in the context of environmental conditions and recording locations.

5:10

4pNS15. Reducing noise from twin supersonic jets using very-low-frequency control. Sandeep R. Murthy (Aerosp., Univ. of Illinois Urbana-Champaign, 306 Talbot Lab., MC-236, 104 S. Wright St., Urbana, IL 61801, srmurth2@illinois.edu) and Daniel J Bodony (Aerosp., Univ. of Illinois Urbana-Champaign, Urbana, IL)

The intense jet noise radiated by closely spaced, twin supersonic hot jets leads to sound-induced structural vibration, fatigue and personnel related operational difficulties. Experimental, theoretical, and computational investigations into the physics and control of jet noise have identified several important sound sources, including wavepackets, screech, Mach wave radiation, and broadband shock associated noise (BBSAN). Reducing the loudest sources of jet noise, without sacrificing propulsive performance, has relied on intuition, parametric survey, or optimal control techniques. With the aim of developing a more general and robust method of jet noise reduction (JNR), we present a physics-based approach that leverages very-low-frequency jet dynamics in order to achieve JNR whilst preserving propulsive performance. Our approach formulates the control problem using the very-low-frequency global modes of the compressible Navier-Stokes operator linearized about the jet mean flow to disrupt the nonmodal transient growth processes. The presentation will showcase uncontrolled and controlled single and twin supersonic hot jets issuing from biconical nozzles, with conditions and geometries motivated by tactical Naval aircraft. The predictions utilize fully resolved numerical simulations whose data inform the control development and which evaluate its performance on the jet exhaust and on its radiated noise.

Session 4pPA

Physical Acoustics, Engineering Acoustics, Structural Acoustics and Vibration, and Underwater Acoustics: Aqueous Acoustic Metamaterials II

Matthew D. Guild, Cochair
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Shane W. Lani, Cochair
Georgia Tech, 1454 Catherine St., Decatur, Georgia 30330

Jason J. Smoker, Cochair
9500 MacArthur Blvd, West Bethesda, Maryland 20817

Invited Papers

1:15

4pPA1. Analysis of an underwater elastic leaky wave antenna. Michael R. Haberman (Walker Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, haberman@arlut.utexas.edu), Kyle Spratt, Michael Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Christina J. Naify (Code 7160, Naval Res. Labs., Washington, DC)

An acoustic Leaky Wave Antenna (LWA) is an electronically simple device that enables frequency-dependent directional sound radiation or reception by coupling a single acoustic transducer to an analogue dispersive waveguide. While most acoustic LWA have focused on airborne acoustic waves, the successful demonstration of acoustic LWA in the underwater environment has been limited to recent numerical and experimental work by the authors [*J. Acoust. Soc. Am.*, **145**(3), 1727 (2019)]. In that work, the LWA was designed through *ad hoc* iteration of finite element (FE) models which is computationally expensive and therefore does not allow for efficient design space exploration. This work will present the derivation of an approximate analytical lumped parameter model for a one-dimensional underwater acoustic LWA consisting of an elastic bar containing periodically arranged mass-in-cavity inclusions. The analytical model employs long-wavelength expansions of sinusoidally forced continuous elements, resulting in a lumped-element approximate model that can be represented with an equivalent circuit. The lumped parameter approximation is compared with FE results and then used to make improvements on the previously reported LWA design.

1:35

4pPA2. Underwater characterisation of acoustic modes supported by structured and unstructured elastic plates. Timothy Starkey (Dept. of Phys. and Astronomy, Univ. of Exeter, Phys. Bldg., Stocker Rd., Exeter, Devon EX4 4QL, United Kingdom, t.a.starkey@exeter.ac.uk), Thomas Graham, Alastair Hibbins, and J Roy Sambles (Dept. of Phys. and Astronomy, Univ. of Exeter, Exeter, Devon, United Kingdom)

Surface waves are supported by unstructured, or appropriately structured, elastic materials. Unstructured plates typically support a number of Surface Acoustic Waves (SAWs) that propagate at the boundary of the elastic materials—these can be bound or leaky. Whilst plates with appropriate structure can support bound Acoustic Surface Waves (ASWs) which propagate in the fluid at the fluid/solid boundary. In this talk, we present the underwater acoustic characterisation of both SAWs and ASWs supported by unstructured and structured aluminium plates. First, we measure the dispersion of the symmetric and antisymmetric leaky Lamb modes of an unstructured aluminium-alloy plate using a broadband ultrasound technique. Beaming of acoustic power over a narrow frequency band is observed. This is attributed to the first order symmetric leaky Lamb mode, which exhibits Negative Group Velocity (NGV) in a small region of its dispersion. Finally, we show that a perforated elastic plate supports a bound ASW. We characterize the dispersion of the supported modes using time-domain measurements of the surface wave pressure recorded as a function of position, and observe acoustic beaming of surface energy over a narrow frequency band. All results show close agreement with numerical models.

1:55

4pPA3. Controlling underwater sound propagation using 3-D-printed phononic crystals. Ahmed Allam (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW (Love Building), Rm. 204, Atlanta, GA 30332-0405, a.allam@gatech.edu), Karim G. Sabra, and Alper Erturk (G. W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

Metamaterial concepts are revolutionizing the way we tackle acoustic wave propagation control. For aqueous media, the realization of metamaterial and phononic crystal concepts using impedance contrast has been mainly limited to using heavy metallic rods which are often bulky and complicated to fabricate to achieve the required geometry. In this work, we introduce a novel and simple technique that utilizes 3-D printing using standard materials such as PLA polymers to construct a phononic crystal which can be engineered to

manipulate underwater waves over a broad frequency range. The proposed design uses an air and polymer matrix to tailor the effective spatial refractive index distribution of the material in order to achieve sufficient refraction. Based on this approach, a 3-D Luneburg gradient index lens prototype was 3-D printed, which can focus underwater acoustic waves with minimal reflection. The cube-shaped lens was fabricated using a common desktop printer and was experimentally characterized. Finite-element modeling of the lens was also performed for band structure analysis and wave field simulations. The results show good agreement between the numerical predictions and the experimental measurements. The developed lens was then used to enhance the performance of an acoustic wireless power transfer system, yielding significant improvement of the power output on the receiver end. The proposed concept and lens design can also be used in other applications such as underwater sensing.

2:15

4pPA4. Study of wave propagation through porous lattices with buckling instabilities in an aqueous environment. Stephanie G. Konarski (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, skonarski@utexas.edu), Christina J. Naify, and Charles Rohde (Acoust. Div., Naval Res. Lab., Washington, DC)

Elastic lattices with cylindrical air-filled voids may undergo significant topology changes when deformed due to the buckling instabilities of the thin elastic ligaments between voids. Such phononic crystals are of interest to, for example, achieve auxetic behavior and create tunable stop bands that vary as a function of the deformation. Most studies reported in the literature by others focus on lattices made of soft elastomers, and the resulting nonlinear deformation or elastic wave propagation due to low-frequency vibrations in air. The current research focuses on transitioning these previously studied concepts on buckling lattices to an aqueous environment. Initial investigations presented here are for 3-D printed lattices comprised of rigid plastic and deformed by leveraging shape memory effects that result in softening and hardening of materials under heating and cooling, respectively. The use of rigid plastic addresses inherent difficulties that arise with soft elastomers, including high loss, fabrication at scales of interest, and appropriate frequency range for wave propagation. The present work focuses on numerical simulations and experimental measurements of wave propagation through finite, plastic lattices with air-filled voids in water. [Work Supported by the Office of Naval Research.]

2:35

4pPA5. Comparison measurements of homogenized material properties of underwater acoustic metamaterials. Benjamin Beck (Appl. Res. Lab, Penn State Univ., PO Box 30, MS 300B, State College, PA 16804-0030, benbeck@psu.edu) and Amanda Hanford (Appl. Res. Lab, Penn State Univ., State College, PA)

A current area of research interest in acoustic metamaterials is the experimental validation of the effective properties computed using analytical and numerical models. This is generally done through an inverse measurement technique, by measuring the reflective and transmitted wave amplitudes and phases to calculate an effective impedance. When the background fluid used for this measurement is air, either an impedance tube or anechoic chamber can be utilized for these measurements. However, if the application of the metamaterial is for in-water applications, the measurement of these properties becomes much more difficult due to many complicating factors. Therefore, there has been little work obtaining homogenized, anisotropic material properties of underwater acoustic metamaterials. In this work, we present comparison of measurement methods of an acoustic metamaterial in an aqueous environment: a water-filled impedance tube and in a reverberant water tank. The metamaterial response will be predicted, and compared to the measurement in the impedance tube and reverberant tank. Additionally, trade-offs between the two experimental methods will be given.

2:55

4pPA6. Design and testing of three-dimensional additively manufactured anisotropic underwater pentamode materials. Colby W. Cushing (Appl. Res. Labs and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, colby.cushing@utexas.edu), Preston S. Wilson, Michael R. Haberman (Appl. Res. Labs and Walker Dept. of Mech. Eng., Univ. of Texas at Austin, Austin, TX), Andrew Norris, and Matthew Kelsten (Dept. of Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

Three-dimensional (3-D) pentamode (PM) materials are used in underwater applications because they can yield impedance matching with water and eliminate shear modes over a wide frequency range while resisting shear deformation in the static limit. Further, a PM lattice can also yield significant anisotropy in sound speed, which is useful for devices that rely on transformation acoustics [Su *et al.*, *J. Acoust. Soc. Am.* 141(6) (2017)]. The present work attempts to verify the predicted anisotropic sound speeds of additively manufactured anisotropic 3-D PM samples. A titanium sample approximately $80 \times 80 \times 60$ mm was suspended in front of a plane-wave source in a water tank to measure the time of flight for wavefronts with and without PM samples present. The measurements were conducted using broadband chirp signals generated by the source, while a scanning hydrophone recorded the response in front of the sample at constant depth. Deconvolution methods were used to extract the impulse response at each scan point and changes in the arrival times of the wavefronts were observed. Frequency domain observations were also made to determine impedance matching characteristics and desired quasi-fluid behavior. Experimental and simulation results will be presented and discussed. [Work Supported by ONR.]

4p THU. PM

Contributed Paper

3:15

4pPA7. Simulating transmission through anisotropic pentamode materials submerged in water. Matthew Kelsten (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, mjk308@scarletmail.rutgers.edu) and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., Piscataway, NJ)

Interest has increased in recent years on the potential for underwater metallic metamaterials, which can be linked to the advancements in metal 3-D printing. One metamaterial of particular interest is an underwater anisotropic pentamode, i.e., a structure designed to support a single longitudinal wave for a broad-range of frequencies whose speed will change depending on its orientation relative to the incident acoustic wave. In an attempt to close the gap between experiment and theory, COMSOL simulations have been developed modeling the transmission and reflection of a normal plane wave incident on an anisotropic pentamode metamaterial consisting of tetrahedral unit cells. The utilization of the Floquet periodic boundary conditions allows for the replication of a semi-infinite medium response composed of

repeating unit cells while only using a finite CAD drawing. Meshing techniques and geometric work-arounds are discussed in an attempt to achieve a cost-effective computation absent of numerical instabilities for a broad range of frequencies. Results show dynamic stress concentration is focused mainly at the joints of the unit cells. Plates designed to separate the fluid medium from the internal structure are shown to exhibit higher modes of vibrations, however contribute little to the far field due to energy flux conservation.

THURSDAY AFTERNOON, 5 DECEMBER 2019 CORONET, 1:00 P.M. TO 2:35 P.M.

Session 4pPPa

Psychological and Physiological Acoustics and Speech Communication: Perceptual Processing of Sound

Mishaela DiNino, Cochair
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Susan R. Bissmeyer, Cochair
Biomedical Engineering, University of Southern California, 24960 Walnut St. Apt. 10, Newhall, California 91321

Chair's Introduction—1:00

Contributed Papers

1:05

4pPPa1. Duplex theory and human localization of 1000-Hz tones. William M. Hartmann (Michigan State Univ., Physics-Astronomy, 567 Wilson Rd., East Lansing, MI 48824, hartmann@pa.msu.edu), Brad Rakerd, Aimee Shore, and Jordan Kassis (Michigan State Univ., East Lansing, MI)

The roles of interaural time difference (ITD) and interaural level difference(ILD) were studied in free-field source localization experiments with 1000-Hzsine tones. Experiments combined real-source trials with virtual-source trialsusing two-channel transaural synthesis based on real-time measurements in the ear canals. Critical virtual trials used fixed, azimuth-independent ILDmagnitudes of 6 or 12 dB. The ILD signs were either consistent with the ITD or opposing. Experiments found that the main role of an opposing ILD was to act as a binary switch, turning on slipped-cycle localization, with some additional sensitivity to ILD magnitude. Therefore, it proved possible to use the localization results from consistent-ILD trials to predict opposing-ILD localization decisions as detailed functions of source azimuth. These results are consistent with the revised duplex theory as observed at lower frequencies[*J. Acoust. Soc. Am.* **139**, 968–985 (2016)], but the effect is more dramatic at 1000 Hz because fewer source azimuths lie in the transition region where slipped-cycle ITD values are large. Parallel headphone experiments with the same listeners showed a similar switching effect in lateralization but also showed greater variability and sensitivity to the ILD itself. [Work supported by the AFOSR.]

1:20

4pPPa2. Laboratory simulations of conversation scenarios: questionnaire results from patient and partner. Peggy B. Nelson (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), Elizabeth Anderson, and Timothy Beechey (Ctr. for Appl. and Translational Sensory Sci., Univ. of Minnesota, Minneapolis, MN)

Hearing-related questionnaires can reveal much about the daily experience of hearing aid users. Nonetheless, results may not fully reflect the lived experience for several reasons, including: users' limited awareness of all communication challenges, limitations of memory, and the subjective nature of reporting. Multiple factors can influence results obtained from questionnaires (Nelson *et al.* ASA Louisville). Consideration of the perspectives of both hearing aid wearers and communication partners may better reflect the challenges of two-way everyday communication. We have developed simulations of challenging conversational scenarios so that clients and their partners can make judgments of sensory aid performance in realistic, but controlled conditions. Listeners with hearing loss and their partners use a client-oriented scale (adapted from the COSI, Dillon, 1997) to report challenging listening conditions such as small group conversations, phone conversations, health reports, and media. Representative scenarios are simulated in the laboratory where clients and partners make ratings of intelligibility, quality, and preference. Results are compared to outcome measures such as the Speech, Spatial and Qualities of Hearing Scale (SSQ, Gatehouse and Noble, 2004) and Social Participation Restrictions Questionnaire (SPaRQ, Heffernan *et al.*, 2018). Results will help refine methods for evaluating the performance of emerging technologies for hearing loss.

1:35

4pPPa3. Sound-source localization when listeners and sound sources rotate: The Auditory Filehne illusion. William Yost (ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu) and M. Torben Pastore (ASU, Troy, NY)

Wallach [*J. Exp. Psychol.* (1940)] pointed out that localizing a sound source in the three-dimensional world depends on an integration of information about auditory-spatial and head-position cues. Yost *et al.* (*JASA*, 2015 and 2019) suggested that a simple sum of the two cues might account for world-centric sound-source localization. Freeman *et al.* [*J. Exp. Psychol.: Hum. Percept. Perform.*(2017)] showed that there is an auditory analog to the visual Filehne illusion (FI). For the auditory FI, a stationary sound source is sometimes perceived as rotating in the opposite direction of head rotation. The perception of a stationary sound source appears to occur when the sound-source also rotates at a rate that is approximately 15% of the head rotation rate. Thus, the integration of auditory-spatial and head-position cues may not be a simple sum. We investigated conditions like the auditory FI for a variety of stimulus and rotation conditions to gain additional insights into how auditory-spatial and head-position cues are integrated for world-centric sound-source localization. [Research supported by grants from NIH (NIDCD), R01DC015214 to WAY; F32DC016808 to MTP; and Facebook Reality Labs to WAY&MTP.]

1:50

4pPPa4. Effect of amplitude fluctuations on the loudness of low-frequency sounds. Carlos Jurado, Darío Gordillo (Escuela de Ingeniería en Sonido y Acustica, Universidad de Las Américas, Quito, Ecuador), and Brian C. Moore (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

The loudness of two-tone complexes (TTCs) with center frequencies (f_c) of 40, 63, 80, and 1000 Hz was matched with that of unmodulated tones (UTs). Frequency differences between the TTC components, corresponding to beat frequencies, f_b, were 1, 2, 5, and 12 Hz. To compensate for the steep decline in hearing sensitivity below 100 Hz, prior to the loudness match, subjects adjusted the relative levels (ΔL) of the TTC components to give maximum beat perception. Twenty-four normal-hearing subjects were tested. The values of ΔL giving best beats were well predicted from the transfer function of the middle ear and the estimated shapes of the auditory filters, assuming that the auditory filter whose output dominated the beat percept was centered somewhat above f_c. At the same root-mean-square level and independent of f_c, TTCs were perceived as louder than UTs for $f_b \leq 2$ Hz, had roughly equal loudness to UTs for $f_b = 5$ Hz, and were less loud than UTs for $f_b = 12$ Hz. The similar pattern observed across f_c suggests that the dependence of loudness on amplitude-fluctuation rate is determined by processes occurring at higher stages of the auditory system than the cochlea.

2:05

4pPPa5. Can pitch rating methods converge on the frequencies within tonal stimuli? Jennifer Lentz (Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, jjlentz@indiana.edu)

This study evaluates whether a pitch rating method (sometimes used to assess tinnitus pitch) converges on the frequencies present in objective tonal stimuli. Twelve normal-hearing subjects completed a pitch rating experiment for seven test stimuli: 6 pure tones ranging from 0.5–12 kHz and two complex tones, with the 0.5-kHz tone used as practice. Subjects listened to 13 different comparison stimuli (presented in random order three times) and rated the similarity of the comparison stimuli to the test stimuli. Comparison stimuli were loudness-matched pure tones ranging from 0.5 to 16 kHz and were presented in an alternating sequence with the test tones. For pure tone test stimuli, results indicated a very high correlation between the highest-rated comparison tone and the test frequency, with slightly less fcorrespondence for the 12-kHz test stimulus. Matches between the ratings of comparison stimuli to complex tone spectra were not as robust, but generally speaking matched in bandwidth. Consequently, this method may be a useful tool for estimating the frequencies present in pure tone stimuli and possibly the bandwidth of complex tones. In comparison to our previous work, we found that both the low-frequency practice stimulus and the alternation of stimuli were critical to obtaining accurate results.

2:20

4pPPa6. Benefits of a smartphone as a remote microphone system. Stephanie Tittle (Behavioral and Brain Sci., Univ. of Texas at Dallas, Dallas, TX), Linda Thibodeau (Behavioral and Brain Sci., Univ. of Texas at Dallas, 800 West Campbell Rd., Richardson, TX 75080, thib@utdallas.edu), Issa M. Panahi (Dept. of Elec. and Comput. Eng., Univ. of Texas at Dallas, Richardson, TX), and Preethi Codanda Chengappa (Behavioral and Brain Sci., Univ. of Texas at Dallas, Dallas, TX)

Remote microphone (RM) systems have been shown to reduce the challenges hearing aid users face with communicating in noisy environments. The RMs worn by the speaker stream their voice wirelessly to the users' hearing aids which results in a significant improvement in the signal-to-noise ratio. Given that the additional cost of a RM may not be feasible for some individuals, the possible use of applications on a smartphone has been explored. The Apple iPhone has an application called 'Live Listen' (LL) that can be used as an RM with made for iPhone hearing aids. The Statistical Signal Processing Research Laboratory at The University of Texas at Dallas has developed an open-source smartphone application for the iPhone that is also designed to be used as an RM. The purpose of this study was to compare the benefit of LL and the open-source app for participants with and without hearing loss on sentence recognition tasks in noise when listening through Starkey Halo2 hearing aids connected to an iPhone 7. Similar improvements in speech recognition were noted for both LL and the open-source app.

Session 4pPPb

Psychological and Physiological Acoustics: Speech and Pitch Perception

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Chair's Introduction—3:00

Contributed Papers

3:05

4pPPb1. Individuals with normal hearing thresholds differ in their use of fine timing cues to identify consonants in noise. Mishaela DiNino (Dept. of Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, mdinino@andrew.cmu.edu), Audra Irvine, Timothy P. Nolan, Barbara Shinn-Cunningham (Neurosci. Inst., Carnegie Mellon Univ., Pittsburgh, PA), and Lori L. Holt (Dept. of Psych., Carnegie Mellon Univ., Pittsburgh, PA)

Noise-induced damage to cochlear synapses (cochlear synaptopathy) weakens encoding of auditory timing in animals and may explain subtle perceptual differences, especially in understanding speech in noise, in humans with normal hearing thresholds. This study utilized a speech-in-noise assessment sensitive to temporal processing deficits to examine the potential link between cochlear synaptopathy and challenges understanding speech in noisy environments. Young adults with normal hearing thresholds identified consonants in /aCa/ utterances in broadband noise that was either: (1) presented from the same simulated spatial location as the speech or (2) spatially separated using a small interaural timing difference (ITD), which forced participants to rely on precise auditory timing differences to achieve spatial release from masking. All listeners performed better with the ITD separation than without it; however, we observed a wide range of ITD cue benefit among participants. These results demonstrate that young adults with normal hearing thresholds vary greatly in their ability to use fine timing cues to spatially separate consonants from noise, which may be one mechanism underlying speech identification impairments in noise among adults with normal hearing thresholds. Ongoing work will determine whether these results correlate with behavioral and physiological measures thought to relate to human cochlear synaptopathy.

3:20

4pPPb2. "Phantom Words" are heard more frequently as coming from the right side of space. Diana Deutsch (Dept. of Psych., Univ. of California, San Diego, 9500 Gilman Dr. #0109, La Jolla, CA 92037, ddeutsch@ucsd.edu), Kevin Dooley (Psych., California State Univ., Dominguez Hills, Carson, CA), and Trevor Henthorn (Dept. of Music, Univ. of California, San Diego, La Jolla, CA)

To experience the "Phantom Words" illusion (Deutsch, 2003) the listener sits in front of two loudspeakers, with one to the left and the other to the right. A sequence is repeatedly presented consisting of two monosyllabic words, or one word composed of two syllables. The sequences coming from both loudspeakers are identical; however, they are offset in time so that when the first sound (word or syllable) is coming from the speaker on the right the second sound is coming from the speaker on the left; and vice versa. Listeners generally perceive different illusory words and phrases, and those appearing as from the right are often different from those appearing as from the left. Here, 20 righthanders and 20 non-righthanders (left-handers and mixed handers) listened to seven such sequences. For each handedness group, ten subjects were seated facing the loudspeakers and ten were seated facing the opposite direction. There was a highly significant tendency for righthanders to hear illusory words and phrases as coming from their right, regardless of their orientation relative to the loudspeakers. Non-righthanders also showed this tendency, but it was less pronounced. Implications for cerebral dominance are here discussed, and illusory sequences are presented as sound examples.

3:35

4pPPb3. Cortical dynamics of word-in-noise recognition. Inyong Choi (Commun. Sci. and Disord., Univ. of Iowa, 250 Hawkins Dr., Iowa City, IA 52242, inyong-choi@uiowa.edu), Subong Kim (Commun. Sci. and Disord., Univ. of Iowa, Iowa City, IA), Adam Schwalje (Dept. of Otolaryngol.-Head and Neck Surgery, Univ. of Iowa Hospitals and Clinics, Iowa City, IA), and Jihwan Woo (Dept. of Biomedical Eng., Univ. of Ulsan, Ulsan, South Korea)

Understanding speech in background noise is a crucial function for communication. Despite the growing body of research on this topic, it is still unexplained how the neural processes for spoken-word recognition are affected by the acoustic degradation of target speech. To address this question, we utilized high-density EEG simultaneously measured during a word identification task with varying levels of background noise. We hypothesize that, if background noise degrades speech sound, listeners will exhibit less immediate processing of speech information, waiting to access lexical entries until most of the word has been heard. We showed that, as we hypothesized, in the more difficult lower-SNR condition, cortical evoked response to target word is weaker in the auditory cortex while the induced and evoked responses from the inferior frontal cortex are delayed but greater, which may imply delayed lexical processing. This study elucidates what neural mechanisms underly effortful speech understanding in challenging acoustic conditions.

3:50

4pPPb4. The cross-domain entrainment effects from pure tones to speech perception. Tzu-Han Cheng (Cognit. Sci., UC, San Diego, 3869 Miramar St., Box 2712, La Jolla, CA 92037, tzcheng@ucsd.edu) and Sarah Creel (Cognit. Sci., UC, San Diego, La Jolla, CA)

Temporal context influences how humans perceive the durations of acoustic events. One of the most influential frameworks, proposed by McAuley and Jones (2003), suggests that entrainment is the underlying mechanism of timing estimation at short time durations. However, their approach has predominantly been tested on simple, music-like acoustic

stimuli (e.g., pure tones). This raises the question of whether the same temporal context mechanisms are operative in other, more complex acoustic domains such as speech sound perception, which some have claimed to operate in modular fashion. In the current study, we first replicated McAuley and Jones (2003) and demonstrated an entrainment effect in our Experiment 1. Based on this finding, we extended the paradigm to the speech domain. In Experiment 2, we tested if an entrainment effect from a series of pure tones could influence the categorization of voiced (/*ba*/) and voiceless (/*pa*/) stop-initial syllables, which were created by varying the durations of voice onset time. Our findings suggest that entrainment from pure tone contexts may influence speech sound categorization. Data from these and ongoing studies has the potential to reveal a general mechanism of short-duration entrainment that can explain temporal context effects on timing perception in acoustically diverse domains.

4:05

4pPPb5. Pitch perception conveyed by cochlear implant and tactile stimulation. Susan R. Bissmeyer (Biomedical Eng., Univ. of Southern California, 1640 Marengo St. HRA 326, Los Angeles, CA 90033, ssubrahm@usc.edu), Juri Hwang (Cinematic Arts, Univ. of Southern California, Los Angeles, CA), and Raymond L. Goldsworthy (Otolaryngol., Univ. of Southern California, Los Angeles, CA)

The healthy auditory system encodes acoustic frequency as a tonotopic and synchronous neural response. Cochlear implants (CIs) excite tonotopic organization through place of stimulation and excite a synchronous neural response through stimulation timing, but typically limited to amplitude modulations less than 300 Hz. The results from two experiments will be presented. The first experiment uses CI stimulation to examine the contributions of place and rate of stimulation to pitch perception. Frequency discrimination thresholds were measured in adult CI users for fundamental frequencies ranging from 100 to 1600 Hz. Stimulation was controlled to probe pitch sensitivity as conveyed by place, rate, and covaried place and rate of stimulation. Subjects received a significant low-frequency benefit from combined stimulation, which diminished above 400 Hz as stimulation rate no longer provided a salient sense of pitch. The second experiment considers how well tactile stimulation can be used to augment pitch perception in CIs. Tactile discrimination thresholds were measured for 55 and 110 Hz. Preliminary results will be presented for the second experiment designed to test how well CI users can combine pitch cues provided by stimulation rate and by tactile stimulation.

4:20

4pPPb6. Extracting beat from a crowd of loosely coupled, concurrent periodic stimuli. Nolan V. Lem (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., 44 Olmsted Rd. Apt. 244, Stanford, CA 94305, nolan.lem@gmail.com) and Takako Fujioka (Ctr. for Comput. Res. in Music and Acoust. (CCRMA), Stanford Univ., Stanford, CA)

In a musical ensemble, performers try to synchronize to a governing tempo by resolving differences in sound-onset timing from individual players even without a conductor's cue. Listeners and players alike must construct an internalized sense of when the beat occurs and adapt to that information dynamically as the performance goes on. Here, we examined this process by simulating individual sound onset timings with an ensemble of 40 virtual "metronomes" around 90 bpm with which we asked listeners to tap along for an approximately 10-beat duration. We manipulated coupling strength at five levels (very-weak, weak, medium, strong, perfect) where stronger coupling corresponds to a more definitively periodic beat. The inter tap interval (ITI) from 8 subjects were analyzed in three segments of the trial duration [early (tap 1–3), middle (4–6), and late (7–9)]. Also, the phase coherence of taps between listeners was compared to the stimulus density. Stronger coupling resulted in more stable ITI, while ITI became shorter in later segments for the medium and weak conditions. Interestingly, taps coincided with the greatest stimulus density for weaker coupling, whereas taps led ahead for stronger coupling. The results suggest that listeners could maintain a collective beat perception but less anticipatorily for less-synchronized sounds.

Session 4pSC

Speech Communication: Speech Perception (Poster Session)

Suzy Ahn, Chair
University of California, Los Angeles, Linguistics, Los Angeles, California 90095

All posters will be on display from 1:15 p.m. to 5:15 p.m. To allow contributors in this session an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:15 p.m. to 3:15 p.m. and authors of even-numbered papers will be at their posters from 3:15 p.m. to 5:15 p.m.

Contributed Papers

4pSC1. Can word frequency norms based on small spoken corpora compete with norms based on popular written corpora? Hayk Abrahamyan (Psych., SUNY at Buffalo, University at Buffalo, Park Hall 204, Buffalo, NY 14260, hayk@buffalo.edu)

Current word frequency norms used in speech research are based on written corpora. Recently, Brysbaert and New (2009) presented newer frequency counts based on larger corpus of film and television subtitles, on the idea that these would more closely approximate actual spoken frequencies. Although Brysbaert and New (2009) showed that their frequencies significantly better predict *visual* reaction time data, they still used corpora with text that was originally written and perhaps heavily edited. In general, most speech researchers prefer using norms from a larger written corpora as norms from spoken corpora are often based on smaller number of tokens (i.e., less then one million). However, many studies use only a small subset of words, frequencies for which may not benefit from a large corpus. Furthermore, many studies dichotomize words into high- and low-frequency groups, thus rendering fine distinctions between frequencies of words computed using a large corpus potentially less useful. The first goal of the current project is to compute word frequency norms using only material from spoken corpora. The second goal is to compare predictions of performance from speech processing experiments of norms from spoken corpora with norms based on popular written corpora. Our preliminary results indicate that for the vast majority of familiar words that are likely to be used in small research studies, any frequency norm even from a small spoken corpus predicts equivalent amount of variance in lexical decision data.

4pSC2. Cognitive load distorts the perception of core acoustic dimensions of speech. Faith Chiu (Dept. of Psych., University of York, York, United Kingdom YO10 5DD, United Kingdom, faith.chiu@york.ac.uk), Lyndon Rakusen, and Sven Mattys (Psych., Univ. of York, York, United Kingdom)

Dual-tasking negatively impacts on speech perception by raising cognitive load. Previous research has shown that cognitive load increases reliance on lexical knowledge and decreases reliance on phonetic detail. Less is known about the effect of cognitive load on the perception of acoustic dimensions below the phonetic level. This study tested the effect of cognitive load on the ability to discriminate differences in duration, intensity, and fundamental frequency of a synthesized vowel. A psychophysical adaptive procedure was used to obtain just noticeable differences (JNDs) on each dimension under load and no load. Load was imposed via N-back tasks at two levels of difficulty (1-back, 2-back) and under two types of load (images, written nonwords). Compared to a control condition with no cognitive load, all N-back conditions increased JNDs across the three dimensions. JNDs were also higher under 2-back than 1-back load. Nonword load was marginally more detrimental than image load for intensity and fundamental frequency discrimination. Overall, the decreased auditory acuity demonstrates that the effect of cognitive load on the listening experience can be traced to distortions in the perception of core auditory dimensions.

4pSC3. The effect of talker accent on the perception of voicing in syllable-initial plosives by American English monolinguals. Peter J. Chong (School of Lang., Literacies & Translation, Universiti Sains Malaysia, Minden, Penang 11800, Malaysia, petercjl88@gmail.com)

A listener's perception of speech is influenced by the talker's characteristics, including their accent. This syllable perception study, which uses a binary forced choice identification task, investigates the effect of talker accent at the level of speech sounds. A single ten-step /ba/-/pa/ continuum varying in voice onset time (VOT) was spliced into identical frame sentences spoken by a native speaker of American English and a non-native speaker whose native language is Mandarin. These frame sentences were modified so that the VOT for the bilabial plosives for both talker conditions is identical. This modification was done to control listeners from using frame sentence VOT as a cue. As the steps in the continuum differed only in terms of VOT for both talker conditions, listeners are only primed by the frame sentence for each item in the experiment. The participants' responses revealed that monolingual American English listeners are more likely to perceive syllables in the native talker condition as /pa/. In particular, the syllable at the 25ms step shows the largest variability between conditions. The results of this study show that listeners can perceive an acoustically identical syllable as being phonemically different when the syllables are spliced into sentences spoken by talkers with different accents.

4pSC4. The role of listening effort during degraded speech recognition: A comparison of the dual-task and pupillometry paradigms. Sarah Colby (Psychol. & Brain Sci., Univ. of Iowa, Seashore Hall, Iowa City, IA 52240, sarah-colby@uiowa.edu) and Bob McMurray (Psychol. & Brain Sci., Univ. of Iowa, Iowa City, IA)

Listening effort is an increasingly important factor in understanding how people recognize speech in adverse conditions, especially for listeners with hearing impairment (Ohlenforst *et al.*, 2017). However, the link between this basic construct and commonly used measures of listening effort is unclear. We compare two measures: dual-task performance and pupillometry, to investigate the relationship between different techniques for measuring effort and performance recognizing vocoded speech. In the dual-task experiment, participants (n = 26) completed a non-linguistic task (pattern matching) either with or without a concurrent speech task. The same participants completed a pupil task, which tracked changes in pupil size while participants recognized words. Both tasks used natural and noise-vocoded speech at two levels of degradation to vary participants' ability to perform accurately. We found no relationship between dual-task performance and task-evoked changes in pupil size. However, the level of degradation uniquely predicted effort in the dual task, while accuracy uniquely

predicted effort in the pupil task. This suggests that different measures of listening effort are sensitive to different underlying factors, including listening conditions and individual ability. Researchers should thus be aware of the sensitivities of a given task when measuring listening effort.

4pSC5. Perceptual adaptation to regional vowel variation. Ellen Dossey (The Ohio State Univ., 108A Ohio Stadium East, 1961 Tuttle Park Pl., Columbus, OH 43210, dossey.1@osu.edu)

Listeners cope with between-talker phonetic variation by adapting their phonemic category boundaries based on evidence from individuals' speech. The current project examined perceptual adaptation to dialect-based variation in the vowels /ɛ/ and /ʌ/, which have lower second formant (F2) frequencies in the Northern dialect of American English relative to the Midland dialect. In a forced-choice lexical identification task, participants identified auditory word tokens which contained vowels along an /ɛ/-/ʌ/ continuum. Words were presented in carrier sentences spoken by either a Northern or Midland talker, and phonemic category boundaries were estimated based on perception of the vowel continuum. Phonemic category boundaries occurred at lower F2s when stimulus words were presented in Northern sentences rather than Midland sentences, indicating perceptual adaptation consistent with the acoustic difference between the dialects. However, this effect was influenced by the phonetic content of the carrier sentence. Adaptation was only consistently found for carrier sentences that included natural tokens of /ɛ/ or /ʌ/, even though all carrier sentences included dialect-specific phonetic features. This suggests that participants were not sensitive to patterns of co-variation among vowels within the two dialects. Rather, they adapted their /ɛ/-/ʌ/ boundary for individual talkers only when provided with direct evidence for that boundary.

4pSC6. Does noise sensitivity mediate physiological measures of listening effort? Alexander L. Francis (Purdue Univ., SLHS, Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, francisa@purdue.edu), Jordan Love, and Mireille Boutin (Purdue Univ., West Lafayette, IN)

Listening to speech in background noise is cognitively demanding and has been shown to engage frontal cortical regions associated with selective attention and cognitive control. However, especially for noise-sensitive individuals, the presence of interfering or distracting noise may also provoke anxiety or frustration, and these same frontal cortical regions also participate in brain networks associated with anxiety and distress. Thus, it is not clear whether previous results showing enhanced frontal activity when listening to speech in noise reflect cognitive effort, noise annoyance, or both. In order to investigate the possibility of dissociating these two aspects of listening to speech in adverse conditions, we will examine the degree to which measures of individual differences in personality traits, noise sensitivity, and cognitive capacities (attention, working memory) predict behavioral and physiological measures associated with listening effort and affective response. Specifically, we will record cardiovascular and electroencephalographic responses to listening to speech in three levels of two types of industrial background noise. The noise types differ only in terms of the relative degree of acoustic properties associated with noise annoyance (tonality, sharpness) instantiated mainly through differences at frequencies not overlapping with those of the speech target.

4pSC7. Do listeners rely on dynamic spectral properties in the recognition of high-pitched vowels? Daniel Friedrichs (Dept. of Theor. and Appl. Linguist, Univ. of Cambridge, Sidgwick Ave., Cambridge CB3 9DA, United Kingdom, daniel.friedrichs@ucl.ac.uk), Francis Nolan (Dept. of Theor. and Appl. Linguist, Univ. of Cambridge, Cambridge, United Kingdom), and Stuart Rosen (Dept. of Speech, Hearing and Phonetic Sci., UCL, London, United Kingdom)

Previous research has demonstrated that intrinsic spectral changes play a significant role in vowel perception. It is possible that dynamic properties are of particular importance at high fundamental frequencies (f_o) where the transfer function of the vocal tract is severely undersampled. The purpose of the present work was to determine whether the phonological function of vowels with very high f_os can be preserved if listeners' identification is based exclusively on static spectral cues. Harmonic additive synthesis was used to generate 324 steady-state versions of the vowels /i y e ø ɛ a o u/ with nine different f_os between 220 and 880 Hz (all 250 ms). The gross spectral shape of each synthesized sound corresponded to a smoothed spectrum derived from a respective template vowel that was naturally produced in isolation by a female native German talker. Native German listeners either identified the spoken or re-synthesized isolated vowels in a two-alternative choice task. Preliminary data showed that listeners' sensitivity (A′) was above chance level in both conditions at all f_os, but significantly higher in the spoken vowel condition. This supports the prediction that dynamic properties enhance listeners' identification at very high f_os, but also indicates that phonological distinctiveness is to some degree preserved at high f_os even when no dynamic spectral information is present. [Work supported by the Swiss National Science Foundation (SNSF) grants P2ZHP1_168375$_{168375}$ and P400PG_180693$_{180693}$.]

4pSC8. Effects of pitch height and contour on duration perception. Emily J. Grabowski (UC Berkeley, 1545 Summit Rd., Berkeley, CA 94708, emily_grabowski@berkeley.edu)

Generally, low tones are produced as longer than high tones and rising as longer than falling (Gandour, 1977). Work in perception has revealed the inverse pattern, which has been described as a reaction to production differences (Yu, 2006). However, while the perception results have been observed reliably cross-linguistically, production patterns appear to have significant variation, which calls into question a direct link between production and perception and motivates more research into the nuances of this relationship. This study investigates the perception of duration in an expanded set of pitch contours via a two-alternative forced-choice task. Subjects were asked to choose which of two same-duration synthesized speech sounded longer, where stimuli varied in pitch height and contour. The results indicate that pitch height has the strongest effect on perceived duration, followed by contour-related effects, which contrasts with predictions based on to Gandour's analysis of the production pattern. Additionally, this study reveals the complexity of the perception pattern beyond the main effects commonly reported in the literature.

4pSC9. Perceptual grouping modulates level-increment discrimination in stop consonant noise bursts. Blas Espinoza-Varas (Commun. Sci. & Disord., OU Health Sci. Ctr., Oklahoma City, OK) and Shaoxuan Guo (Dept. of Surgery, OU Health Sci. Ctr., 800 Stanton L. Young Blvd., Oklahoma City, OK 73126-0901, shaoxuan-guo@ouhsc.edu)

Does perceptual grouping modulate the ability to discriminate level increments (ΔL) in stop-consonant noise bursts? In consonant-vowel-consonant (**CvC**) words comprising an ≈80-dB vowel (**v**), a pre-vocalic (**Cv**) and a post-vocalic (**vC**) stop-consonant noise burst (≈ 60-dB SPL), we measured ΔL thresholds for bursts in isolation or in **CvC** context. Each interval of 2I-2AFC tasks presented a **CvC** word (e.g., /pæk/ /pæk/), and participants had to discern ΔL in **Cv** or **vC**. Relying on linguistic labels, the auditory events are perceived as two auditory objects (**Cv**-v-**vC** and **Cv**-v-**vC)** that group bursts and vowels together, hampering attention to ΔL. To discern ΔL in **Cv** or **vC**, events must be reorganized into three auditory objects: the to-be-attended pre-vocalic (**Cv**-**Cv**) or post-vocalic burst pair (**vC**-**vC**), and the to-be-ignored vowel pair (**v**-**v**). Relative to bursts in isolation, bursts in context produced higher thresholds (context effect), which depended on the time separation between the bursts and the vowel: lower for the object apart from (post-vocalic) than for the object adjoining (pre-vocalic) the vowel (temporal-position effect). In addition to being robust and persistent, these effects are relatively general, evincing in tasks with one or two observation intervals, with or without variability in the temporal position of ΔL.

4pSC10. F0 as a secondary cue for consonant aspiration in Mandarin Chinese. Yuting Guo (English, George Mason Univ., 4400 University Dr., 3E4, Fairfax, VA 22030, yguo16@gmu.edu) and Harim Kwon (English, George Mason Univ., Fairfax, VA)

Previous studies on stop production in Mandarin Chinese have reported conflicting results on consonant-induced F0 perturbation (e.g., Xu and Xu, 2003, Luo, 2018). This study investigates whether Mandarin listeners use F0 information as a cue for consonant aspiration when F0 perturbation is

limited and inconsistent in production. Specifically, we examine Mandarin listeners' perception of phonological /t^h/-/t/ contrast, focusing on how voice onset time (VOT) and post-stop F0 influence their perceptual judgments. Mandarin listeners heard /tu/~/t^hu/ syllables with the initial stop co-varying in its VOT and post-stop F0 in the four lexical tone contexts, and were asked to identify the syllable they heard. Results demonstrate Mandarin listeners relied mainly on VOT for /t^h/-/t/ contrast, but the effect of VOT interacted with post-stop F0 and lexical tone. F0 did not influence the perceptual judgments when VOT was unambiguous. However, when VOT was ambiguous, Mandarin listeners identified the stops with high F0 as /t^h/, and those with low F0 as /t/. Lexical tones mediated this effect: Tones with low pitch onset elicited more unaspirated responses than those with high pitch onset. These findings suggest that Mandarin listeners use F0 as a secondary cue for consonant aspiration, but to different extent in different lexical tones.

4pSC11. Speaking clearly improves speech segmentation in optimal listening conditions. Zhe-chen Guo (Linguist, The Univ. of Texas at Austin, 307 E 31st St. Apt. 105, Austin, TX 78705, y9024131@gmail.com) and Rajka Smiljanic (Linguist, The Univ. of Texas at Austin, Austin, TX)

Speaking clearly improves intelligibility particularly in noisy environments—a benefit that may be partly attributed to enhanced ease of speech segmentation. However, speech segmentation is facilitated by coarticulation (Fernandes *et al*., 2007), which may be reduced in hyper-articulated clear speech (Scarborough and Zellou, 2013). This study investigated the effect of clear speech on speech segmentation in quiet and in noise for native and non-native listeners with an artificial language (AL) learning experiment. Six trisyllabic AL words (e.g., /kutupi/) were produced by a native English speaker both clearly and conversationally. Six speech streams were created, each containing 20 pitch-flattened tokens of each AL word concatenated in random order without intervening pauses. In the learning phase, participants heard either conversational or clear AL speech streams in quiet or masked with speech-shaped noise. Next, in a two-alternative forced-choice test, participants identified the AL words spoken in quiet and in the style they were exposed to. Preliminary results showed that clear speech improved segmentation relative to conversational speech for native English listeners in quiet only. The results suggest that the use of exaggerated information about segment identity aids speech segmentation in optimal listening conditions but is disrupted under perceptual load.

4pSC12. Auditory feedback control of vocal intensity during speech and sustained-vowel production. Allison I. Hilger (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Unit C, Evanston, IL 60208, allisonhilger2020@u.northwestern.edu), Sam Levant, Jason Kim (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL), Rosemary A. Lester-Smith (Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX), and Charles R. Larson (Commun. Sci. and Disord., Northwestern Univ., Evanston, IL)

Vocal intensity is an important speech component for communicative effectiveness, such as speaking above ambient noise in a loud restaurant, and also for communicative appropriateness, such as speaking quietly in a library. Speakers adjust their vocal intensity using information from their auditory feedback of how loud they perceive their speech. In this study, we were interested in the control of auditory feedback for vocal intensity. Participants repeatedly produced both a simple, sustained-vowel (i.e., "ahhh") and a short target phrase (i.e., "You know Nina?") while their voice auditory feedback was briefly perturbed up or down in loudness. We then measured their reflexive intensity response produced quickly after the perturbation to correct for the unexpected change in loudness. In a similar study on pitch perturbations, speakers produced larger reflexive pitch responses as a function of perturbation magnitude in phrase production but did not produce this differential pattern in sustained vowel production (Chen *et al*., 2007). This result suggests that auditory feedback is more sensitive to changes in suprasegmental features of speech when produced with linguistic intent. In our current study, we are interested in whether this same pattern will be observed for control of vocal intensity.

4pSC13. What does cross-linguistic perception tell us about the phonetics-phonology interface? Phil J. Howson (Univ. of Oregon, Sidney Smith Hall, 4th Fl. 100 St. George St., Toronto, M5S 3G3, philh@uoregon.edu) and Irfana M (Netaji Subhash Chandra Bose Medical College, Jabalpur, India)

The internal representation of linguistic information has been the subject of much debate. Recent work has shown that brain activation in response to different categories of segments, such as obstruents or approximants, triggers different neural networks. This work examines the perception of English and Malayalam native speakers' perception of seven fricatives, /f, s, ʂ, ʃ, ɕ, x, h/, to examine what the acoustic-perceptual grouping of segments within a class reveals about internal representation. The results indicated distinct grouping of the fricatives into three categories, anterior /f/, sibilant /s, ʂ, ʃ, ɕ/, and posterior /x, h/. We also observed an expansion in the perceptual space for sibilant fricatives as a function of perceiver L1. We suggest the distinct clustering in the perceptual space reflects the internalized representation of these segments. Segment groupings are formed on the basis that they share class features, supporting the notion that class membership is dictated, at least in part, by acoustic-perceptual relationships. We suggest that the expansion is related to language exposure specific to Malayalam, which has three sibilant fricatives in its inventory, /s, ʂ, ɕ/. The result of the exposure is an expansion and more well-defined category boundaries for segments in the native inventory.

4pSC14. Detection of nasals as a function of duration. Ting Huang (Haskins Labs., Graduate Inst. of Linguist, Rm. B306, HSS Bldg., No. 101, Section 2, Kuang-Fu Rd., Hsinchu City 30013, Taiwan, funting.huang@gmail.com)

Previous studies have shown distributional restrictions on nasal stops. Specifically, in many languages nasal stops are followed only by a nasal vowel, which is also usually comparatively longer to contrast with the vocalic context adjacent to oral stops. This was observed as a phonological constraint in Taiwanese (Southern Min), but not in French. This study tests the discriminability of Taiwanese and French listeners on plain and nasal stops. The speech stimuli were cross spliced nasal versus oral stops and oral versus nasal vowels: CV, NV, CṼ, NṼ. The vowel duration was also manipulated to see if the perceived nasality will be enhanced when the vowels are longer, and further affect the intelligibility for the consonants before them. Preliminary results indicated that the Taiwanese listeners predicted wrong consonantal categories when the nasality of the following vowel is not agree with the preceding consonant. French listeners showed no such effect. The durational effect shows the opposite: Taiwanese listeners perceived the same stop category regardless of the duration of the following vowel, whereas French listeners are biased toward the nasal category when the following vowel is longer.

4pSC15. Cue weighting in the perception of Tagalog stress. Hyun Kyung Hwang (Univ. of Tsukuba, Kyonan-Cho 2-9-21-1903, Musashino, Tokyo 180-0023, Japan, hwang.kyung.gu@u.tsukuba.ac.jp), Naonori Nagaya (The Univ. of Tokyo, Mitaka, Japan), and Julián Villegas (Univ. of Aizu, Aizu Wakamtsu, Fukushima, Japan)

Relative importance of different acoustic cues in perceived prominence is language specific. In Tagalog, it has been argued that there is a lexical level stress, and a stressed syllable exhibits higher fundamental frequency, longer duration, and stronger intensity (Gonzales, 1970). Still, the relative weight of these cues has not been discussed in the prosodic literature. The purpose of the current study is to quantify the relative weighting of multiple acoustic cues in the perception of lexical stress in Tagalog. A forced-choice identification test was conducted with a disyllabic word /babad/. Each of the three acoustic cues in both syllables was manipulated in two levels, i.e., strong and weak, under fully crossed combinations of the cues. A total of sixty-four stimuli were tested in four sessions, and fifty-three native speakers of Tagalog participated in the test. The result of a generalized linear model using a binomial distribution revealed that the primary perceptual cue is duration. Longer duration was significantly associated with perceived prominence, and higher fundamental frequency to a lesser degree. However, intensity cues did not seem to play a significant role in stress perception in Tagalog.

4pSC16. Effects of variability and internal structure on malleability of phonetic categories: A case of [i] versus [u]. Reiko Kataoka (Linguist and Lang. Development, San José State Univ., One Washington Square, San Jose, CA 95192-0093, reiko.kataoka@sjsu.edu) and Hahn Koo (Linguist and Lang. Development, San José State Univ., San Jose, CA)

Recent studies on retuning of phonetic categories by lexically guided perceptual learning suggest an inverse relationship between inherent acoustic variability and malleability of phonetic categories—the greater the variability, the smaller the retuning effect. However, it remains to be investigated what perceptual processing mechanisms are responsible for the observed relationship. Extending our previous study (Kataoka and Koo, 2017) that compared the degree of malleability between [u] (more variable) and [i] (less variable), we not only compared the size of retuning effect between the two vowels but also examined how listeners judge category goodness of synthesized stimuli from a [i]-[u] continuum. Our subjects (1) showed signs of category retuning for [i] but not for [u] and (2) judged goodness of stimuli in a more gradient manner and took longer to do so when asked to judge with reference to [u] than [i]. The results suggest that the two vowels differ in terms of their acoustic variability as well as their internal structure and that relative difficulty in resolving input speech signal in reference to a category such as [u] might be one reason the category is less malleable than a less variable category such as [i].

4pSC17. Exploring the influence of intonational structure on perception of contrastive vowel length in Tokyo Japanese. Hironori Katsuda (Linguist, UCLA, 3130 Sawtelle Blvd. 204, Los Angeles, CA 90066, katsuda1123@gmail.com) and Jeremy Steffman (Linguist, UCLA, Los Angeles, CA)

A recent topic of interest in the literature is how intonational/prosodic structure influences speech perception. We test how perception of a phonemic vowel length contrast in Japanese is influenced by the implied prosodic position of a target sound in a carrier phrase, with the target drawn from a vowel duration continuum. Based on previous findings in English [Steffman, *JASA* (2019)], we predicted that listeners should require *longer* vowel duration to categorize a vowel as phonemically long when it is in phrase-final position, reflecting an expectation of phrase-final lengthening. We manipulated implied position by changing only pitch in a carrier phrase, as informed by Japanese intonational phonology. The carrier phrase was otherwise identical across conditions. Additionally, psychoacoustic effects of pitch on perceived duration are unlikely to influence categorization as predicted by the prosodic account, making this a good test case for prosodic patterns' independent influence in speech perception and the processing of durational cues. Results will be discussed in terms other recent findings related to intonation and perception of duration, and implications for the interface of prosody and speech perception more generally.

4pSC18. Gender differences in voice onset time perception. Seung Kyung Kim (UCSD, 5 Ave. Pasteur, Aix en Provence 13100, France, kim.seungkyung@gmail.com)

It has been known for a while that a gender difference in voice onset time (VOT) production exists such that males typically produce shorter VOTs than females (Swarts, 1992; Robb *et al*., 2005). However, potential gender differences in VOT perception have not been systematically investigated. This study addresses this question and asks if male and female listeners of English exhibit a different crossover point in a VOT continuum between a voiceless and voiced sound. 11-step VOT continua between /t/ and /d/ (VOT ranges from 70 ms to 5 ms) were created and a set of identification tasks were performed. Results show female listeners switch from a /t/-percept to a /d/-percept at an earlier VOT step (i.e., with a longer VOT) than male listeners do. That is, when female listeners start to categorize a token as a /d/-sound, male listeners still categorize the same token as a /t/-sound. This difference in the perceptual pattern between males and females mirrors their difference in production. VOTs for /t/ are typically shorter for male than female speakers, and male listeners categorize shorter VOT tokens as /t/ more often than female listeners do. This finding highlights the role of social factors like gender in the production-perception link.

4pSC19. The role of between- versus within-speaker acoustic variability in vocal identity perception. Jody E. Kreiman (UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90403, jkreiman@ucla.edu) and Yoonjeong Lee (UCLA, Los Angeles, CA)

Our recent studies [*JASA* **145**(Pt. 2), 1930, (2019); this conference] show that acoustic spaces characterizing within- and between-speaker variability in voice quality have similar structures, with a few features (acoustic variability and formant dispersion) important for all speakers combined with idiosyncratic features characterizing individual talkers. These findings suggest that voice discrimination should be based on shared features, while "telling voices together" should depend on knowledge of each individual's vocal idiosyncrasies. To test this hypothesis, we selected a set of voices that varied systematically in acoustic closeness to each other and to the center of the group acoustic space, based on values of shared features. Listeners were presented with multiple samples of each voice and were asked to sort the voices into piles according to perceived speaker identity. Based on prototype models of voice perception, we hypothesized that errors in telling voices apart would be strongly predictable from distances in the group acoustic space, but that errors in telling voices together would not be significantly associated with these distances. Separate analyses will attempt to shed light on the features and strategies involved in this second kind of judgment. [Work supported by NIH/NSF.]

4pSC20. The influence of word size and tonal sequence probability on Mandarin well-formedness judgments. 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, Bellingham, WA), and Julie Liss (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Work in tonal languages has demonstrated that distributional information is important in speech processing. Probability of syllable + tone combinations play a role in speech processing in Mandarin (Wiener and Ito, 2015, 2016; Wiener and Turnbull, 2013), and transitional probabilities of tone-bearing vowels are important in Cantonese segmentation (Gomez *et al*., 2018). However, the importance of variables like word size and tonal sequence probability is less well understood. Further, while disyllabic words are by far the most common word size in Mandarin (Duanmu, 2007), research into word-likeness has only explored listeners' judgments of single-syllable prompts (Kirby and Yu, 2007; Myers and Chen, 2017; Gong and Zhang, 2019). The role of word length and distributional probabilities of tonal sequences within multi-syllabic words need to be examined. We conducted a phonological well-formedness task varying stimuli by number of syllables and tonal sequence probability, hypothesizing that listeners would rate two-syllable words and words with higher probability tonal sequences as better formed than low probability word forms. The experiment is ongoing, but preliminary findings fail to indicate a significant effect of either word size or tonal sequence probability on well-formedness ratings. Current analyses explore the potential influence of lexical neighborhood on listeners' judgments.

4pSC21. The effects of high variability training on voice identity learning. Nadine Lavan (Dept. of Speech, Hearing & Phonetic Sci., Univ. College London, 2 Wakefield St., London WC1N 1PF, United Kingdom, n.lavan@ucl.ac.uk), Sarah Knight, Valerie Hazan, and Carolyn McGettigan (Dept. of Speech, Hearing & Phonetic Sci., Univ. College London, London, United Kingdom)

High-variability training has been shown to benefit the learning of phonetic contrasts and new face identities. Here, we investigated whether high-variability training also aids voice identity learning. In Experiment 1, we contrasted high variability training (including stimuli extracted from several different recording sessions and speaking styles) with low variability training (including stimuli extracted from one recording session of read sentences; i.e., one speaking style). Listeners learned to recognise 4 voice identities (2 through high-variability training, 2 through low-variability training) and were subsequently tested on an old/new recognition task using novel read sentences. We found no high-variability training advantage in Experiment 1 – instead we found a high-variability disadvantage. However, in this experiment there was full overlap of speaking style in the low

variability voices across the two task phases. We therefore ran a second experiment (Experiment 2), in which test stimuli were sourced from a previously unheard speaking style: The manipulation of high versus low-variability training was otherwise achieved in the same way as before (many speaking styles versus one speaking style). In this experiment, we observed a high-variability advantage. Findings are discussed in the context of the mechanisms thought to underpin advantages for high-variability training.

4pSC22. Comparative function of Spanish vowels and consonants in lexical and socioindexical processing tasks using auditory discrimination. Stephanie Lain (Cowell Academic Services, Univ. of California Santa Cruz, Santa Cruz, CA 95064, lain.steph@ucsc.edu)

A series of speech perception experiments involving lexical decision and talker identification tasks shows that Spanish-speaking listeners rely heavily on consonants to identify words and vowels to identify talkers. The study replicates a similar study for English. Participants were 94 native Spanish-speaking undergraduate students who performed a same-different auditory discrimination (forced choice) task to detect a word (Version A) or a speaker (Version B). Listeners heard pairs of isolated nonsense words produced by speakers of Venezuelan Spanish. In Experiment 1, a control group heard unaltered stimuli. Experiments 2 and 3 followed the same design was but with vowels excised (Experiment 2) or consonants excised (Experiment 3). Results showed that overall discrimination was best with the unaltered stimuli (Experiment 1, as predicted), followed by vowel-only stimuli (Experiment 3). Furthermore, it was shown that listeners relied more heavily on consonants to perform the lexical decision task relative to the talker identification task (Experiments 2 and 3). These findings confirm results from previous studies using English-speaking participants, showing a minimal role for crosslinguistic variation. Current results contribute to the body of knowledge on speech universals and have implications for the dual functions of lexical and socioindexical processing across languages.

4pSC23. Factors shaping vowel perception biases in adults. Ying Ying Liu (School of Commun. Sci. & Disord., McGill Univ., 2001 McGill College Ave. - 8th Fl., Montreal, PQ H3A 1G1, Canada, ying.y.liu@mail.mcgill.ca), Matthew Masapollo (Boston Univ., Boston, MA), Lucie Menard (Univ. of PQ at Montreal, Montreal, PQ, Canada), and Linda Polka (School of Commun. Sci. & Disord., McGill Univ., Montreal, PQ, Canada)

Vowel discrimination is often asymmetric such that discriminating the same vowel pair is easier in one direction compared to the opposite direction. The Natural Referent Vowel framework interprets these directional asymmetries as a universal bias favoring "focal" vowels (i.e., vowels with prominent spectral peaks formed by the convergence of adjacent formants). The Native Language Magnet model instead interprets asymmetries in terms of a language-specific bias due to distortion of perceptual space around native language vowel prototypes. To test these competing views, Masapollo *et al*. (2017) compared English- and French-speaking adults' discrimination of synthetic /u/ variants; this was informative because English /u/ is naturally less focal than French /u/. Their findings revealed asymmetries to be predicted by focalization only; although stimulus limitations may explain the lack of prototype effects. Here, we synthesized a more refined series of vowel stimuli systematically varying in smaller psychophysical steps around the English /u/ and French /u/ prototypes to augment the measurement of focalization and prototype effects. Native English speakers completed a category goodness-rating task followed by an AX-discrimination task using these new variants. Results indicated effects of both focalization and prototype, which are moderated by the size of acoustic intervals along the stimulus series.

4pSC24. Does dynamic visual information in talking faces influence the perceptual restoration of phonemes? Julia Irwin (Haskins Labs., New Haven, CT), Alyssa Lotto, Kayleigh Ryherd (Dept. of Psych., Southern Connecticut State Univ., New Haven, CT), and Matthew Masapollo (Dept. of Speech, Lang. and Hearing Sci., Boston Univ., 677 Beacon St., Boston, MA 02215, mmasapol@bu.edu)

It is well-established that the perception of speech can be highly influenced by visible articulatory information. Recently, Irwin *et al*. (2017) demonstrated a robust effect in which visual speech cues perceptually "restore" a speech sound that has been acoustically weakened. Here we investigated the nature of the visual information that elicits this perceptual illusion. To accomplish this, we utilized an oddball paradigm in which perceivers were presented with acoustic /ba/ (the more frequently occurring standard stimulus) and /a/ tokens (the infrequently presented deviant stimulus). The acoustic tokens were dubbed with three types of video tokens: (1) A full face articulating /ba/; (2) the same face articulating /ba/ but with the oral-facial region pixelated; or (3) a point-light facial display of the produced /ba/ that depicted the isolated kinematics of the visible lip movements. Results indicated that perceivers showed visual phonemic restoration (reduced accuracy in detecting deviant /a/) in the presence of the natural talking face, but not in the presence of either the pixelated or schematic (point-light) faces. These findings suggest that the extracted kinematic information may not be sufficient to elicit the restoration effect, or that isolated kinematic cues do not integrate with acoustic speech in a robust manner.

4pSC25. Perceptual evaluation of speech naturalness in speakers of varying gender identities. Brandon Merritt (Speech and Hearing Sci., Indiana University-Bloomington, 945 Basswood Circle, Bloomington, IN 47403, bmmerrit@iu.edu) and Tessa Bent (Speech and Hearing Sci., Indiana University-Bloomington, Bloomington, IN)

Most auditory-perceptual evaluations of speech from individuals who are transgender have focused on perceived masculinity and femininity. Recent work has begun to investigate speech naturalness, which many transgender individuals report as a voice training goal. Voice masculinity/femininity may be linked to perceived speech naturalness based on listener sensitivity to talker gender and expectations surrounding prototypical male and female voices. Because transgender speakers are often rated as sounding less masculine or feminine than cisgender speakers, their perceived speech naturalness may be negatively impacted. This study investigates how speech naturalness ratings relate to masculinity-femininity ratings and gender identification (accuracy and reaction time). Spontaneous speech samples from 20 transgender (10 transmasculine; 10 transfeminine) and 20 cisgender (10 male; 10 female) speakers were included in three tasks: a two-alternative forced-choice gender identification task, a speech naturalness rating task, and a masculinity/femininity rating task. Utterances rated as less natural took longer to identify and were identified less accurately in the gender identification task; furthermore, they were placed in the middle of the masculinity/femininity scale. These results suggest that training to align a speaker's voice with their gender identity may concurrently improve perceptual speech naturalness.

4pSC26. Contextual effects on the distinction of Japanese length contrasts in reverberation. Eri Osawa (Information and Commun. Sci., Sophia Univ., Tokyo 102-8554, Japan, eri1989.16.11@gmail.com) and Takayuki Arai (Information and Commun. Sci., Sophia Univ., Tokyo, Japan)

Temporal structure in a word is crucial for understanding speech. In the distinction between Japanese singleton and geminate stops, listeners tended to perceive a consonant as geminate when the preceding vowel was longer [Ofuka *et al*., *J. Phonetics Soc. Japan* **9**(2), 59–65 (2005)]. At the same time, listeners may not be able to use the contextual information in reverberation since the boundaries between each speech sound may become blurred by reverberation elongating and overlapping the sounds. The current study investigated whether the category boundary between singleton and geminate might change depending on duration of the preceding vowel in reverberation. Participants made a choice between singleton and geminate consonants embedded in an /a a/ context. The duration of the preceding vowel was changed. The experiments were conducted in a reverberant (RT = 2.6 s) and non-reverberant (RT = 0.1 s) condition. The results showed a significant shift in category boundary which indicated that listeners gave more "geminate" responses when the preceding vowel was longer, especially in the reverberant condition. The preceding vowel lengthened by reverberation caused listeners to choose "geminate." Listeners might be more sensitive to the length of the adjacent phonemes for assessing the degree of reverberation.

4pSC27. Lexically dependent estimation of acoustic information in speech III: Cross-splicing verification of cue weights. Charles Redmon (Linguist, Univ. of Kansas, 1541 Lilac Ln., Rm 427, Lawrence, KS 66046, redmon@ku.edu) and Allard Jongman (Linguist, Univ. of Kansas, Lawrence, KS)

In two prior studies (Redmon and Jongman, 2018, 2019, *JASA*), results of open- and closed-class word recognition tasks on items drawn from a large single-speaker database (Tucker *et al*., 2018) were used to train a model where acoustic cue weights were optimized to distinguish words in the lexicon, rather than a balanced inventory of phones. From that work, cues were identified that had a greater weight when considering the lexicon as a whole than when studying a symmetric set of contrasts in controlled syllable productions. To verify the causal role of such cues in word recognition, two new cross-splicing versions of the open- and closed-class tasks were run with a subset of items in each prior experiment. In each, an *enhancement* condition was created by cross-splicing diphones from items in the database with more distinct values on a given cue dimension, and consequently greater model-predicted accuracy. For each such item, a parallel *reduction* condition was created wherein accuracy was predicted to decrease due to the cross-splicing of a diphone with a more ambiguous value of a given cue. Results will serve to validate the relative role of the different cues in a manner that is external to the model-fitting procedure.

4pSC28. Phonetic cues influence judgment of syntax. Emily Remirez (Linguist, UC Berkeley, 1203 Dwinelle Hall #2650, Berkeley, CA 94720, eremirez@berkeley.edu)

This speech perception experiment provides evidence of interaction between phonetic and syntactic cues associated with socio-/ethnolects. These phonetic and syntactic (socio)linguistic cues were crossed in stimuli recorded by speakers of African American English (AAE), Southern Standard British English (BrE), and General American English (GAE). Thus, some of the sentences listeners heard contain 'matching' cues—for example, a *have raising* question with BrE phonetic features—while the others contain "mismatching" cues much less likely to have been experienced together by listeners. Listeners (n = 30) rated the spoken utterances' acceptability on a scale of 1 to 5; response time was recorded. The mixed-effects linear regression fitted to the Z-transformed data shows BrE and AAE constructions are more acceptable when the phonetic cues match. The effect is most robust for the BrE accent; compared to estimates for the GAE accent (β = 0:9538, p < 0.001), those for the BrE accent are overall lower (β = −0.106, p = 0.048). The penalty is effectively counteracted, however, when the BrE voice is producing the associated syntactic construction (β = 0:181, p = 0:017). Response time data also suggests stimuli with matching cues are processed faster (t = -2:3586, p = 0:01856).To my knowledge, this is the first experimental evidence of an effect of accent on syntactic acceptability.

4pSC29. Processing prosodically driven variations in lexical access. Seulgi Shin (Linguist, Univ. of Kansas, 1846 Tennessee St., Apt. #4, Lawrence, KS 66044, seulgi.shin@ku.edu) and Annie Tremblay (Linguist, Univ. of Kansas, Lawrence, KS)

This study investigates whether prosodically driven variants constrain lexical access by focusing on the processing of prosodically licit and illicit Korean lenis plosives and nasals. Lenis plosives are aspirated phrase-initially but voiced phrase-medially; nasals undergo nasal weakening ("denasalization") phrase-initially but not phrase-medially. Importantly, denasalized nasals share acoustic similarities with voiced lenis plosives. Seoul Korean listeners completed four cross-modal priming experiments where primes in phrase-initial sentential position were related (experimental) (e.g., /tapal/ "bundle") or unrelated (control) (e.g., /tʃodɛ/ "invitation") to visual targets (e.g., 다발/tapal/). The target began with a lenis plosive (Exps. 1 and 2) or nasal (Exps. 3 and 4). The experimental prime began with a prosodically licit (Exp. 1: aspirated lenis, Exp. 3: denasalized nasal) or illicit (Exp. 2: voiced lenis, Exp. 4: nasalized nasal) variant. The results showed that experimental primes with a prosodically licit variant (Exps. 1 and 3) facilitated word recognition. For prosodically illicit variants, experimental primes with a nasalized nasal (Exp. 4) facilitated word recognition but experimental primes with a voiced plosive did not (Exp. 2). The facilitation in Exp. 4 suggests that prosodically driven variants do not constrain lexical access; the lack of facilitation in Exp. 2 is attributed to lexical competition from nasal-initial words.

4pSC30. Background music contributes to error in the operating room. Alexandra L. Bruder (Vanderbilt Univ. Medical Ctr., Nashville, TN), Kendall Burdick (Univ. of Massachusetts Med. School, Nashville, TN), Russ Beebe (Vanderbilt Univ. Medical Ctr., Nashville, TN), Clayton Rothwell (Biomedical Informatics, The Ohio State Univ. College of Medicine, Columbus, OH), and Joseph Schlesinger (Anesthesiology Critical Care Medicine, Vanderbilt Univ. Medical Ctr., 1211 21st Ave. South, Medical Arts Bldg., #526, Nashville, TN 37212, joseph.j.schlesinger@vanderbilt.edu)

Music and alarms have both been shown to improve performance in surgeons in the operating room (OR), but may interfere with anesthesiologist communication and performance. This experiment investigated the effects of music and alarms on speech intelligibility in a cognitively demanding multitask setting using simulated clinical, speech intelligibility, and visual vigilance tasks. Anesthesiology residents' (n = 25) speech intelligibility performance was measured using the Coordinate Response Measure (CRM) in a 2 × 2 within-subjects design that varied background (simulated OR noise or OR noise with music, both normalized to 60 dB SPL) and alarm type (conventional or a novel auditory icon conforming to IEC 60601-1-8). Linear mixed-effects models showed music had a significant reduction in accuracy (p < 0.001), increased RT (p < 0.001), more instances of no response (p < 0.001). With music, average RT was increased by ∼0.2 s (95% CI: 0.07, 0.39), the odds of a correct response were reduced by ∼39% (95% CI: 25, 50), and the odds of no response were increased by 340% (95% CI: 246, 467). There was no evidence that alarm type impacted CRM performance. A device that modulates music volume during critical phases could reduce communication interference.

4pSC31. A perceptual perspective on Cantonese tonal mergers-in-progress using lexical categorization. Rachel Soo (Linguist, Univ. of Br. Columbia, 100 St. George St., Toronto, ON M5S 3G3, Canada, soorache@gmail.com) and Molly E. Babel (Linguist, Univ. of Br. Columbia, Vancouver, BC, Canada)

Cantonese is generally described as having a six tone system, composed of 3 level, 2 rising, and 1 falling tones. Researchers have observed that several of these tones have been merging; for example, Tone (T) 2 and T5 [Bauer *et al*., *LVC* **15**(2), 211–225 (2003)], T3 and T6, and T4 and T6 [Mok *et al*., *LVC* **25**(3), 341–370 (2013)]. In perception, discrimination-based paradigms show that Cantonese speakers are slower and poorer at discriminating said merging tone pairs [Mok *et al*., *LVC* **25**(3), 341–370 (2013); Soo and Monahan, *BLS* **43**(2), 47–54 (2017)]. In this study, we examine these mergers in terms of word identification to develop an understanding of the nature of the mergers. Homeland and heritage Cantonese listeners categorize minimal pairs on an 11-step continuum for T2–T5, T3–T6, and T4–T6. Pictures are used as the lexical endpoints, as many participants in the sample do not read characters. Analysis of the categorization functions provide further empirical data on whether listeners are perceptually merged in a way that affects lexical identification. Additionally, these data will shed light on the taxonomy of the mergers to determine whether these are mergers by expansion, approximation, or transfer.

4pSC32. Influences of preceding speech rate and prosodic position on listeners' perception of durational cues. Jeremy Steffman (Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095-1543, jsteffman@g.ucla.edu)

Research suggests that listeners incorporate expectations about the temporal organization of prosodic structure in their perception of durational cues. Steffman (2019, *JASA*) tested sensitivity to phrase-final lengthening: listeners categorized a "coat"∼"code" vowel duration continuum (increased duration cues voicing) in a carrier phrase. The target was either phrase-medial or phrase-final. Listeners required longer vowel durations for a "code" response when the target was phrase-final, suggesting an expectation of lengthening in final position influenced perception of duration. The present study tests how listeners are sensitive to preceding speech rate changes in tandem with prosodic/positional effects. Using the same stimuli as Steffman (2019), preceding speech rate was manipulated orthogonally to position (2×2): a target was phrase-final or medial, preceded by either slow or fast precursor speech. Results show independent influences of rate and position on categorization, which are differently localized. Positional effects are concentrated at longer vowel durations which align with typical durations for phrase-final vowels. Rate effects localize in the middle region of the continuum. These results offer some support for the idea the prosodic effects in perception depend on durational patterns in the language, and extend Steffman (2019) to show that listeners integrate both speech rate and prosodic information in perception.

4pSC33. Biphone probability and neighborhood density effects on phonetic categorization. Jeremy Steffman (Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095-1543, jsteffman@g.ucla.edu) and Megha Sundara (Linguist, UCLA, Los Angeles, CA)

Adults' categorization of speech sounds is influenced by different aspects of lexical and phonological organization. These include neighborhood density and biphone probability. The role of both these influences in models of speech recognition has been a topic of debate. This is in part because these two effects are difficult to disambiguate given the measures are highly correlated: denser neighborhoods tend to have high biphone probability sequences. Accordingly, interactive models can account for neighborhood density effects based on lexical feedback, with biphone probability effects as their by-product. Conversely, in the absence of feedback from the lexicon, autonomous models can explain density effects using biphone probabilities alone. We present two experiments testing cases which disambiguate these effects. In Experiment 1, listeners categorized re-synthesized /ɛ/~/æ/ continua. Vowel continua were presented in CVC frames, both endpoints of which were English non-words. Crucially, we manipulated the neighborhood density and biphone probabilities of each endpoint of the continuum independently. Our results show an independent contribution of neighborhood density and biphone probability in categorization. In Experiment 2, we are now using eye-tracking to disentangle the time course of each effect. Results will be discussed in the context of the role of feedback in speech recognition.

4pSC34. Judgments of American male talkers who are perceived to sound gay or heterosexual: Certain social contexts don't affect perception of personality traits. Erik C. Tracy (Psych., Univ. of North Carolina Pembroke, PO Box 1510, Pembroke, NC 28372, erik.tracy@uncp.edu)

Researchers proposed that different social contexts influence how listeners perceive talkers' emotional states (Bachorowski and Owren, 2002). Furthermore, it was discovered that, in the absence of any context (i.e., listeners were not informed of talkers' sexual orientation), some personality traits (e.g., confident, mad, stuck-up, and outgoing) were associated with gay-sounding male talkers, while other personality traits (e.g., boring, old, and sad) were associated with heterosexual-sounding male talkers (Tracy, 2016). The first experiment examined whether a stronger association between these traits and voices might emerge if listeners were informed of the talkers' sexual orientation (i.e., greater social context). After hearing a spoken utterance, listeners rated gay-sounding and heterosexual-sounding talkers along the previously mentioned personality traits. For the majority of traits, the results demonstrated that listeners' personality judgments did not change if they were told or were not told the sexual orientation of the talker. The second experiment revealed that, in the absence of vocal information, participants do not associate these personality traits with these particular talkers. Thus, the data from the two studies align. If listeners have knowledge of a talker's sexual orientation, they do not associate this knowledge with particular personality traits and, thus, their personality judgments of the voices would not be affected.

4pSC35. The role of semantic predictability in adaptation to non-native-accented speech. Kayla Walker (Dept. of Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, kwalker3@uoregon.edu), Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, Eugene, OR), and Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., Bloomington, IN)

Non-native-accented speech is more difficult for native listeners to understand than native-accented speech. However, listeners can improve their abilities to understand nonnative—accented speech through exposure and training. The goal of this project is to explore whether exposing native listeners to different sentence types affects listeners' adaptation to non-native speech. Listeners will be trained on high predictability sentences (e.g., "The color of a lemon is yellow"), low predictability sentences (e.g., "Mom said that it is yellow"), or semantically anomalous sentences (e.g., "The green week did the page"). Previous research has demonstrated that semantic predictability impacts speech perception, but its influence on adaptation to nonnative speech is unknown. Will training with low predictability or anomalous stimuli require listeners to focus more attention on the acoustic-phonetic properties of the accent and thus lead to greater adaptation and generalizable learning? Or will training with high predictability stimuli provide valuable semantic information that will allow listeners to create a better framework for improving perception? The data from this experiment will shed light on perceptual mechanisms, including how semantic predictability interacts with adaptation and learning.

4pSC36. Temporal cues from visual information benefit speech perception in noise. Yi Yuan (Speech, Lang., and Hearing Sci., Univ. of Florida, 1600 SW Archer Rd., Rm. D2-77, Gainesville, FL 32610, yiyuan56@ufl.edu), Andrew Lotto, and Yonghee Oh (Speech, Lang., and Hearing Sci., Univ. of Florida, Gainesville, FL)

Speech perception in noise is a challenge for older adults and listeners with hearing loss. It is also challenging for a subset of listeners with normal hearing thresholds. One way to improve performance in noisy settings is to provide visual information of the talker. The current study examines what aspects of this visual input are necessary to receive an audio-visual benefit. To test the hypothesis that a visual analogue of the speech amplitude envelope aids speech comprehension, speech sentences in babble noises, with or without visual stimuli, were presented to listeners. In this case, the visual analogue was a sphere that varies based on normal, mismatched, and reversed relation to the amplitude envelope of the speech signal. Even when the audio-visual correlation is reversed (higher amplitude = smaller volumes), a significant improvement in speech perception in the auditory-visual condition versus the audio-only condition was gained even though no visual representation of phonetic information was available. These results provide strong evidence that the amplitude envelope can be inferred from visual displays and can be integrated online in speech perception. This study has implications for potential technological enhancements to speech perception with hearing devices – in particular, the integration of a non-auditory signal.

4pSC37. Two case studies of perceived age, perceived speech quality, and speech acoustics. Eric J. Hunter (Dept. of Commun. Sci. & Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, ejhunter@msu.edu), Sarah H. Ferguson (Dept. of Commun. Sci. & Disord., Univ. of Utah, Salt Lake City, UT), and Lady Catherine Cantor Cutiva (Dept. of Collective Health, Universidad Nacional de Colombia, East Lansing, MI)

Physiological processes involving speech anatomical structures change with age and affect voice and speech production. Additionally, those with high voice demand (e.g., singers, teachers, coaches) have also demonstrated voice and speech production related changes. From a listener's perspective, speech changes can be used to give insight into the vocal health and the age

of a talker. This study compared perceived production quality, perceived age, and speech acoustics of two elderly individuals. Speech samples were used from two public figures. The first person (F, 78 y/old) gave an 8-h continuous public address. The second person (M, 49-98 y/old), spoke publicly for nearly 50 years. Using speech samples from both individuals, the following analysis were completed: (1) direct age estimation of samples; (2) production quality ratings trained listeners rated the samples voice quality using the GRBAS; and (3) acoustic analysis of the samples was performed. As would be expected, actual age and estimated age were related for the longitudinal samples representing 50 years of aging. Also, speech production quality and speech acoustic characteristics also changed with age. Interestingly, analyses of estimated vocal age was shown to decrease with prolonged vocal speaking showed a negative relationship (younger estimated age after hours of speaking).

THURSDAY AFTERNOON, 5 DECEMBER 2019 REGENT, 1:15 P.M. TO 4:45 P.M.

Session 4pSP

Signal Processing in Acoustics: Array Signal Processing II

Erin Fischell, Chair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 266 Woods Hole Rd., WHOI, MS 11, Woods Hole, Massachusetts 02543

Contributed Papers

1:15

4pSP1. Improving the dominant mode rejection beamformer with median filtering. David C. Anchieta (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, danchieta@umassd.edu) and John R. Buck (ECE, Univ. of Massachusetts Dartmouth, North Dartmouth, MA)

Adaptive beamformers (ABFs) outperform conventional beamformers in detecting weak sources while attenuating background noise and strong interferers. Changing environments limit the number of snapshots that an ABF can average to estimate the sample covariance matrix (SCM). The dominant mode rejection (DMR) beamformer [Abraham and Owsley, *Oceans* (1990)] overcomes rank-deficient SCMs by imposing a structured covariance matrix which replaces the noise subspace eigenvalues by their average. However, the DMR beamformer often overestimates the dominant subspace dimension to avoid interferers contaminating the beamformer output. Overestimating the dominant subspace dimension introduces a bias to the background noise power estimate. This bias impairs the array gain and output signal to interferer and noise ratio (SINR). This talk proposes a modification to the DMR, replacing the average of the noise eigenvalues with an estimate of background noise power derived from the median of all eigenvalues. Simulations demonstrated that this new median-DMR beamformer improves the output SINR by up to 0.9dB compared to the standard DMR beamformer in scenarios which conservatively overestimated the dominant subspace dimension and suffered from a rank-deficient SCM. [Research supported by ONR 321US.]

1:30

4pSP2. Frequency-wavenumber spectrum estimation using blended dominant mode rejection beamforming. Kathleen E. Wage (George Mason Univ., 4400 University Dr., MSN 1G5, Fairfax, VA 22030, kwage@gmu.edu)

Capon [*Proc. IEEE* (1969)] designed the minimum variance distortionless response (MVDR) beamformer to obtain spectral estimates with better resolution than the conventional averaged-periodogram estimator. The MVDR spectrum is a function of the inverse of the sample covariance matrix (SCM), which often must be regularized prior to inversion. To address conditioning problems, Abraham and Owsley [*IEEE Oceans* (1990)] developed a modified MVDR approach called dominant mode rejection (DMR). The DMR beamformer defines its weights using a structured covariance consisting of a low-rank interference subspace plus an orthogonal noise subspace. It assumes the rank of the interference is known. Recently, Buck and Singer [*IEEE SAM* (2018)] proposed the blended DMR beamformer that eliminates the need for rank estimation by defining a weight vector that is an affine combination of fixed-rank DMR beamformers. This talk investigates frequency-wavenumber estimation using blended DMR, focusing particularly on efficient implementations for large linear arrays. Adapting the approach described by Therrien [Prentice Hall (1992)] for MVDR, the blended DMR spectrum for an equally spaced array can be computed using fast Fourier transforms of the sample eigenvectors. Results will be illustrated using experimental data from underwater vertical arrays. [Work supported by ONR.]

1:45

4pSP3. High-amplitude focusing of ultrasound in air using time reversal. Carla B. Wallace (Dept. of Phys. & Astron., Brigham Young Univ., N283 ESC, Provo, UT 84602, carlabutts2.718@gmail.com) and Brian E. Anderson (Phys. & Astronomy, Brigham Young Univ., Provo, UT)

In this study, time reversal (TR) signal processing methods are used to create a focus of airborne ultrasound (between approximately 30 kHz and 50 kHz) in a room. There are unique challenges presented by ultrasonic sources, such as the highly directional nature of the sources, and attenuation of ultrasound in air. The aim of this study is to create a TR focus whose amplitude is as high as possible using the clipping processing method. Because ultrasonic sources tend to be very directional, this study explores the impact of using methods to make the sources more omnidirectional, which increases the number of reflective paths that contribute to the TR focus. This study explores various configurations of source and microphone in the room to determine which configuration yields the highest amplitude. Because thermoviscous loses for ultrasound are higher than for audible frequencies, a smaller room is used in the experiments to increase focal amplitudes.

2:00

4pSP4. Comparison of sonar systems for mapping of seaweed and infrastructure on macroalgae farms. Erin Fischell (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS 11, Woods Hole, MA 02543, efischell@whoi.edu), Kevin Manganini, Sean Whelan, Amy Kukulya, Timothy K. Stanton, and Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

To increase production of seaweed to the point where it is economically viable as a biofuel, many developments are required in areas such as breeding and offshore farming systems. To inform decision making by farmers, maps of features that describe farm infrastructure, growth of seaweed, and fish populations are needed. As a part of the Advanced Research Projects Agency-Energy (ARPA-E) MARINER (Macroalgae Research Inspiring Novel Energy Resources) program, acoustic systems are being assessed for quantitative autonomous mapping. Multiple data collection sequences including various sonars and autonomous vehicles were used to better understand the efficacy of different sonars for this application. In data collection trials in New England sugar kelp farms using an autonomous surface vehicle, side-by-side sonar data were collected using a broadband split-beam sonar system, a low-cost 120 kHz narrow-band system, and a low-cost recreational fish finder. An autonomous underwater vehicle carrying a side-scan sonar system and broadband scientific split-beam sonar system was further used to collect data on both sugar kelp in New England and giant kelp in California. Measurements are compared for different mission objectives.

2:15

4pSP5. Using the frequency-difference autoproduct to passively range remote sources in the deep ocean. 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)

Signal processing techniques utilizing the nonlinear frequency-difference acoustic-field autoproduct have been applied in recent years to successfully localize remote sources in the shallow ocean [Worthmann *et al.*, *JASA* **138**, 3549–3562 (2015)] and in the deep ocean [Geroski and Dowling, *JASA* (to appear)] by switching the signal processing to below the signal band frequencies. The successful source localization results obtained here indicates the robustness of frequency differencing techniques to the problem of mismatch between measured and computed acoustic fields that plagues source localization algorithms based on correlating computed and measured field. The promising deep ocean results were obtained by analyzing PhilSea10 experimental acoustic recordings from a water-column spanning vertical array with 149 receivers and an approximate length of 5 km. Given that such an extensive receiving array is unlikely to be routinely available, an investigation was conducted to determine how reduced array aperture and transducer count influence source ranging results from frequency-differencing techniques for 100-Hz bandwidth signals having center frequencies from 170 and 260 Hz, and source to array ranges between 100 and 500 km. Ranging results from this study are presented for both simulated and measured in-band signals, and are calculated using single-digit Hz difference frequencies. [Sponsored by ONR.]

2:30

4pSP6. Recursive least squares calibration and tomographic imaging with an ultrasonic transducer array. Robert W. Adams (Houston Res. Ctr., Aramco Services Co., 17155 Park Row Dr., Houston, TX 77084, robert.adams@aramcoservices.com), Tim Thiel, Jonathan Harrist, and Max Deffenbaugh (Houston Res. Ctr., Aramco Services Co., Houston, TX)

Travel time measurements from an acoustic array interrogating a fluid within the cross-section of a pipe include systematic errors: from mispositioning of array elements, group delays from the transceiver circuits at each element, and acoustic propagation effects between transmitter/receiver pairs. An enhanced calibration model is described which includes the influence of these error sources, and a recursive least squares solution is applied to find calibration travel-time corrections that minimize systematic errors. This enhanced calibration model is applied *in-situ* to an acoustic transducer array, resulting in increased accuracy and precision of acoustic travel-time measurements. Tomographic images of fluid velocities are derived from these travel-time measurements. The fluid velocities are mapped to known velocities of individual fluid phases, resulting in tomographic images of multiphase flows.

2:45

4pSP7. Comparison and real-time implementation of fixed and adaptive beamformers for speech enhancement on smartphones for hearing aid study. Nikhil Shankar (Elec. and Comput. Eng., Univ. of Texas at Dallas, 800 W Campbell Rd, Richardson, TX 75080, nxs162330@utdallas.edu), Gautam Shreedhar Bhat, and Issa M. Panahi (Elec. and Comput. Eng., Univ. of Texas at Dallas, Richardson, TX)

In this paper, we compare the performance of fixed and adaptive beamformers as an application to speech enhancement (SE) algorithm. The proposed signal processing pipeline works in real-time for hearing aid devices using a smartphone as an assistive tool. The proposed method consists of a Wiener filter based single-channel SE method along with beamformer as a pre-filter. In this work, multiple beamforming approaches like delay and sum, minimum variance distortionless response (MVDR) and generalized sidelobe canceller (GSC) beamformers are considered and compared. The use of beamformers is shown to improve the signal to noise ratio (SNR) in real-world noisy conditions through the results presented in this paper. Objective, intelligibility evaluation, and subjective test results show the comparison and efficiency of the proposed method at different SNR levels.

3:00–3:15 Break

3:15

4pSP8. Reproduction and analysis of near and far stereophonic sound fields with compact uniform linear arrays. Elliot Patros (Music, Univ. of California San Diego, 3531 Ray St., San Diego, CA 92104, epatros@ucsd.edu), Tahereh Afghah (Music, Univ. of California San Diego, La Jolla, CA), and Peter Otto (Dysonics, Inc., San Diego, CA)

Stereo speakers are commonly used for multi-modal listening. This includes both near field, on-axis or "sweet-spot" listening; as well as far field, off-axis or "room-fill" listening modes. Meanwhile, compact uniform linear arrays (cULA) are increasingly available as an alternative to conventional stereo systems. However, acoustic properties of cULA limit their ability to reproduce accurate sound fields for both modes simultaneously. This paper proposes several modifications to wave field synthesis that serve together as a simple and robust method for multi-modal stereo reproduction with cULA. Near field performance is analyzed by comparing estimates of stereophonic localization for both the proposed system and conventional stereo. Far field performance is analyzed by comparing magnitude responses across both systems' sound fields. Finally, connections between array configuration parameters and stereophonic reproduction artifacts are highlighted.

3:30

4pSP9. Acoustic source localization using drone-embedded microphone array. Mateusz Guzik (The Faculty of Mech. Eng. and Robotics, AGH Univ. of Sci. and Technol., Adama Mickiewicza 30, Kraków 30-059, Poland, mateusz.guzik0@gmail.com), Konrad Kowalczyk, Szymon Woźniak (Faculty of Comput. Sci., Electronics and Telecommunications, AGH Univ. of Sci. and Technol., Kraków, Poland), Mieszko Fraś, Klara Juros, Daniel Kaczor (The Faculty of Mech. Eng. and Robotics, AGH Univ. of Sci. and Technol., Kraków, Poland), and Piotr Walas (Faculty of Comput. Sci., Electronics and Telecommunications, AGH Univ. of Sci. and Technol., Kraków, Poland)

High mobility and an ability of gathering data from large terrains makes Unmanned Aerial Vehicles (UAVs) an excellent platform for placing visual or acoustic sensors. One recently emerging application of UAVs is search and rescue operation, during which drones are used to localize people in distress. A common approach to determine the target position is to rely on visual data recorded by cameras. However, in situations of limited visibility such as in presence of smoke, at night or when a person is trapped under debris, acoustic information can be exploited to perform the localization of people in distress. Solutions based on acoustic information gathered by drone-embedded microphone array are a promising alternative to the methods based on vision, and they are currently being widely examined for UAV applications. The main issues encountered in acoustic source localization using drones include high ego-noise and wind produced by the propellers. This paper investigates the statistical properties of drone's ego-noise and proposes an algorithm for acoustic source localization which exploits the sparsity of sound sources in time-frequency domain. A comparison of the results obtained by the proposed method and by commonly used approaches clearly shows the benefits of using the proposed processing.

3:45

4pSP10. Experimental study on source direction finding using a cylindrical array considering 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 (Sejong Univ., Seoul, South Korea), Yeon-Seong Choo (Korea Univ. of Sci. and Technol., Daejeon, South Korea), and Sung-Hoon Byun (Korea Res. Inst. of Ships and Ocean Eng., Daejeon, South Korea)

A circular array is widely used for source direction finding due to its more compact configuration and smaller angular dependency compared to a linear array. In underwater environments, however, high drag forces make it difficult to utilize the circular array especially for the case where there is high-speed ocean current or the platform with the array is moving. To overcome this operational difficulty an array of hydrophones flush mounted on a cylindrical structure may be applied. A recent numerical study shows that the beamforming performance is improved by considering the scattered sound field by a cylinder. Recently, a tank experiment using an array of eight receiver element on a cylindrical structure has been done for underwater source direction finding. In the experiment narrowband and wideband signals are used for the sound transmitter. The experimental results also show that the lower side-lobes are obtained when the scattering effect is considered during beamforming analysis. [This work is financially supported by the research project PES3180 funded by KRISO.]

4:00

4pSP11. An algebraic geometry approach to passive source localization. Margaret Cheney (Mathematics and ECE, Colorado State Univ., Math, 1874 Campus Deliver, Fort Collins, CO 80523, cheney@math.colostate.edu) and Ivars P. Kirsteins (NUWC, Newport, RI)

We examine the application of algebraic geometry-based localization methods to 3-D underwater acoustic source localization. An approach to passive source localization that is commonly used in the electromagnetics community is to first determine the time difference of arrivals (TDOAs) by cross-correlating pairs of received signals, and then from the TDOAs use algebraic geometric methods to determine the source location. In particular, a TDOA for a receiver pair determines a hyperboloid on which the source must lie. Consequently the source must lie at the intersection of the TDOA hyperboloids. Because hyperboloids are examples of polynomials, systems of hyperboloids can be solved with numerical algebraic geometry software such as Bertini or Macaulay2. Such software typically works by starting with a reference system of polynomials whose exact solution is known, and tracking the solution while gradually deforming the reference system into the desired one. This approach is less well-known in the sonar community because the sonar wave propagation environment is more complicated. We begin by applying this algebraic geometry approach to the case of an iso-velocity range-independent sound speed environment and then investigate extensions of this approach to underwater waveguides with non-homogeneous sound speed profiles.

4:15

4pSP12. Performance analysis of sparse and small arrays. Tsih C. Yang (College of Information Sci. and Elec. Eng., 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. To improve performance, either high resolution beamforming is used, which has its own problems, or one needs to increase the array aperture, hence requiring more elements on the array. To reduce the cost, sparse and/or nested arrays have been proposed, such as the coprime arrays, to achieve approximately the same aperture, and hence the same beam resolution with a less number of elements. For sonar arrays, deconvolution is shown to yield an unique solution as opposed to, say, image processing. For coprime array, deconvolution algorithm is shown to yield a narrower beam width, lower sidelobe levels and higher array gain than the product or min processing, Deconvolution is shown to yield superdirectivity and supergain for a small array. This raises new research issues about optimal array configuration design.

4:30

4pSP13. Adapting underwater acoustic communication networks to changing oceanic conditions Using opportunistic multipath signaling schemes. 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)

A fundamental bottleneck to high data-rate shallow water acoustic communications is rapidly fluctuating delay spread, which is caused by highly unpredictable multipath scattering by the moving sea surface, sea bottom as well as fluid motion. A rich literature exists on how to track and mitigate the mutipath interference and thus recover the direct arrival between the transmitter and receiver. This talk will review some of the recent methods proposed, in particular light of different representations of the channel multipath in delay, time, frequency domains and combinations thereof. The talk will also present how the different representations of channel multipath can be exploited to design opportunistic signaling schemes that exploit rather that mitigate multipath effects. We will also discuss the impact of such opportunistic signaling, particularly in the context of diversity strategies in communication networks.

Session 4pUW

Underwater Acoustics and Animal Bioacoustics: Ship Source Level Estimation: Methods and Measurements

Dag Tollefsen, Cochair
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David E. Hannay, Cochair
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Chair's Introduction—1:00

Invited Papers

1:05

4pUW1. Practical ship noise level measurements in shallow water. Kai A. Abrahamsen (Noise & Vib., DNV GL, Veritasveien 1, Høvik, Akershus 1363, Norway, kai.abrahamsen@dnvgl.com)

Ship noise radiation has been recognized as an important environmental factor which attracts increasing attention. It is important to be able to measure the noise radiation in order to control ship noise and to evaluate consequences of ship generated noise. Operation of ships is associated with significant costs. Large cruise vessels may have operational costs in the range 10–20 000.-$ per hour. Smaller merchant vessels are less expensive, but the operational costs are still an important factor. Hence, it is important to be able to perform underwater noise measurements fast at or near the sailing area of the vessel using mobile equipment. Underwater noise measurements can be rather complex if all theoretical influences are to be controlled, but may be simplified if considered in a practical way and still yield acceptable accuracy. This talk will describe practical ways to perform underwater noise measurements, discuss factors leading to variability in the measurements and describe how to limit the variability as much as possible while still keeping the measurements within a simple practical frame.

1:25

4pUW2. Container ship source level and directionality measurements. Martin Gassmann (European Patent Office, The Hague, The Netherlands, mgassmann@ucsd.edu), Sean M. Wiggins (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA), Lee B. Kindberg (Maersk Line, Charlotte, NC), and John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Underwater radiated noise from container ships was measured opportunistically from multiple spatial aspects to estimate directionality and signature source levels. Transiting ships were tracked via the Automatic Identification System in a shipping lane while acoustic pressure was measured at the ships' keel and beam aspects. Port and starboard beam aspects were 15, 30, and 45 deg in compliance with ship noise measurements standards [ANSI/ASA S12.64 (2009) and ISO 17208-1 (2016)]. Source levels were derived with a spherical propagation (surface-affected) or a modified Lloyd's mirror model to account for interference from surface reflections (surface-corrected). Ship source depths were estimated from spectral differences between measurements at different beam aspects. In addition, recordings were made at a ~10 deg starboard aspect and utilized to measure and compare signatures of MAERSK G-class container ships before and after the ships were equipped with new propellers as part of MAERSK's $100+ million Radical Retrofit Program.

1:45

4pUW3. A multivariate analysis of vessel source levels from the enhancing Cetacean habitat observation (ECHO) ship noise database. Alexander O. MacGillivray (JASCO Appl. Sci., 2305–4464 Markham St., Victoria, BC V8Z7X8, Canada, alex@jasco.com), David E. Hannay, and Heloise Frouin-Mouy (JASCO Appl. Sci., Victoria, BC, Canada)

The Vancouver Fraser Port Authority's Enhancing Cetacean Habitat Observation (ECHO) program acquired a large database of several thousand systematic commercial ship noise measurements between September 2015 and April 2017. These measurements were used to develop a multi-variate linear regression model of vessel source levels against several parameters that describe vessels and their measurement conditions. Covariates in the multi-variate model included ship category, ship length, dead-weight-tonnage, static draught, effective wind speed magnitude and direction, ship speed, and surface angle. The regression analysis examined the statistical

significance of each covariate's regression coefficient and produced a set of fit coefficients, one for each covariate for each frequency band. The multi-variate model was used to normalize the measurements for each category, and the remaining data variance reflected vessel-specific differences in noise emissions that could not be attributed to measurement circumstances. The multivariate analysis produced a powerful ship noise model that can predict decidecade band monopole source level (MSL) and radiated noise level (RNL), which is useful for understanding noise emissions variations with ship characteristics and under different operating conditions. [This research was funded by Transport Canada.]

2:05

4pUW4. Merchant vessel source level estimation from low-frequency vector sensor systems. Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Bldg 232, Rm. 114, Monterey, CA 93943, kbsmith@nps.edu), Paul Leary (Dept. of Phys., Naval Postgrad. School, Monterey, CA), and Thomas J. Deal (Naval Undersea Warfare Ctr., Newport, RI)

Low-frequency acoustic vector sensor systems deployed in propagation favorable locations have the potential to not only provide accurate tracking of distant merchant vessels but also, with a combination of AIS data and propagation modeling, produce good estimates of ship source levels. In this work, data from two similar low-frequency acoustic vector sensor systems will be examined using this approach. One system collected data approximately 3 km off the coast of Big Sur, California at the edge of the shelf break, while the other system collected data from a depth of about 900 m near the mouth of the Monterey Bay Canyon. Directional and temporal variations in the ambient noise field will also 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 periods 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. The results from each system will be compared with the goal of reducing model uncertainty.

Contributed Papers

2:25

4pUW5. Vector acoustic study of ship noise during the 2017 Sediment Characterization Experiment off New England. 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 bottom deployed system measuring particle velocity and pressure. Results using IVAR in the Sediment Characterization Experiment (SBCEX) conducted off New England (spring 2017), involving active sources have been presented [P. H. Dahl and D. R. Dall'Osto, *IEEE J. Ocean. Eng.* (2019)]. Here, passive ship noise is studied from a 200-m length cargo vessel that is tracked over a 10 km course a 15 knots for which the closest point of approach (CPA) to the IVAR deployment location (1.25 m above the seafloor) was 500 m, as confirmed by Automatic Identification System (AIS) data. The time-frequency interference pattern of the noise as the vessel closes and opens in range is studied by way of vector acoustic field indicators. These are non-dimensional ratios of second-order quantities, e.g., kinetic energy over potential energy, which preserve magnitude over the track. Inferences on the seabed made from study of the field indicators are compared with geoacoustic models emerging from studies related to SBCEX. The aspect-dependent source level of the vessel is also estimated, with method tested on a known, omni-directional source towed past IVAR from the research vessel R/V *Endeavor*. [Study supported by Office of Naval Research.]

2:40

4pUW6. International standards for the measurement of underwater noise from vessels: Numerical modelling in support of a shallow water standard. Victor F. Humphrey (ISVR, Univ. of Southhampton, Highfield, Southampton SO17 1BJ, United Kingdom, vh@isvr.soton.ac.uk), Yin Cen (ISVR, Univ. of Southhampton, Southampton, United Kingdom), Stephen P. Robinson, and Lian Wang (NPL, Teddington, United Kingdom)

ANSI and ISO have issued standards for the measurement of the radiated noise level of surface vessels in deep water. In addition, a draft standard for the conversion of such measurements into a monopole source level, as required for environmental noise models, is at an advanced stage. However, many operators and investigators are constrained to making measurements in shallow water where both sea-surface and seabed effects are important. In order to study the influence of these effects a numerical study has been performed using both an image source model and OASES, with very good agreement. The goal of this work has been to understand the variability of the correction factor required to convert measurements of received pressure, at short range, into the equivalent source level in such an environment. These calculations have included the effects of reflection at both the sea-surface and an attenuating seabed. The influences of the water depth, seabed type, distance of closest approach, and number and location of the hydrophones have been investigated. The results show how the correction factor varies and indicate the extent to which simplified approximations can be used for the correction factor.

2:55

4pUW7. Variability of radiated underwater noise measurements for a small research vessel in shallow water. Victor F. Humphrey (ISVR, Univ Southhampton, Highfield, Southampton SO17 1BJ, United Kingdom, vh@isvr.soton.ac.uk) and Alex Brooker (Clarke Saunders Acoust., Winchester, Hampshire, United Kingdom)

The increased interest in the potential environmental impact of noise from shipping is resulting in the development of measurement methods for determining the radiated noise level and equivalent monopole source level of vessels. It is important to understand the random and systematic uncertainties associated with such techniques. The EU SONIC project provided an opportunity to make repeated measurements on the University of Newcastle research vessel, the Princess Royal, in a shallow water environment (100 m deep) over a number of days. Two multiple hydrophone arrays operated by two of the project partners, CETENA and the University of Southampton, were deployed from the same moored support vessel. This data is reviewed to illustrate the variation associated with such measurements and the impact of using multiple hydrophones and multiple measurement runs on the uncertainty in the radiated noise level and the calculated source level. The results are also compared with a later measurements of the same vessel in a different shallow water environment (20 m deep) using bottom moored hydrophones.

3:10

4pUW8. Source level predictions of surface ships using seabed characterization in the New England Mudpatch from viscous grain shearing model. David P. Knobles (Knobles Sci. and Anal., PO Box 27200, Austin, TX 78755, dpknobles@yahoo.com), Preston S. Wilson (Mech. Eng. and ARL, Univ. of Texas at Austin, Austin, TX), William S. Hodgkiss, Michael J. Buckingham (Scripps Inst. of Oceanogr., La Jolla, CA), Tracianne B. Neilsen (Phys., Brigham Young Univ., Provo, UT), Lin Wan, and Mohsen Badiey (Univ. of Delaware, Newark, DE)

Acoustic inferences from remote sensing of surface ship radiated noise in bottom-limited ocean environments are difficult due to a generally unknown geoacoustic structure of the seabed. The idea here is to first *calibrate* an acoustic sensing system by estimating geoacoustic parameter values by utilizing controlled sources with known levels. Then, the acoustic field that results from a surface ship is utilized to estimate the ship source levels where the transmission loss is constructed based on the previous estimate of the seabed parameterization and measurements of the sound speed profile at the time that the ship noise measurements were made. This concept was tested with acoustic data collected on two vertical line arrays (VLAs) during the Seabed Characterization Experiment in the New England Mudpatch during April 2017, in 75 m of water. The acoustic field was produced with a controlled source radiating in the 1.5–4 kHz band. A maximum entropy method provided estimates of viscous grain shearing (VGS) model parameter values. Then, using this VGS parameterization, ship source levels as a function of aspect and frequency out to about 3 kHz were extracted for the RV Endeavor and several merchant ships passing near the VLAs. [Work supported by ONR N00014-16-C-3065.]

3:25–3:40 Break

3:40

4pUW9. Ship source level estimation in an uncertain environment via trans-dimensional Bayesian inversion. Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Boks 115, Horten 3191, Norway, dag.tollefsen@ffi.no), David P. Knobles (KSA LLC, Austin, TX), Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada), and William S. Hodgkiss (Scripps Inst. of Oceanogr., La Jolla, CA)

This paper considers the joint estimation of ship source spectral levels and environment parameters of a layered seabed via a trans-dimensional Bayesian matched-field inversion approach, with applications to shallow-water data collected with acoustic arrays 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 ship source levels that include uncertainty due to the environment and source depth/range. Approaches to modeling of a distributed source (i.e., as multiple point sources) in the inversions will also be considered. The approach is applied to low-frequency tonal (propeller and machinery) noise in the 10–300 Hz frequency band due to large container ships passing near the arrays.

3:55

4pUW10. Individual horizontal array elements as windows to estimating ship radiated noise signature lobing patterns. Dugald Thomson (Dept of Oceanogr., Dalhousie Univ., Rm. 3635 - 1355 Oxford St., Halifax, NS B3H 4R2, Canada, dugald@dal.ca) and David R. Barclay (Oceanogr., Dalhousie Univ., Halifax, NS, Canada)

A horizontal hydrophone array is typically beamformed to produce the directivity required to produce array gain and determine bearing to a contact of interest. For ships passing near an array each omnidirectional hydrophone also provides a number of discrete “look windows” at the ship from a range of received angles. By assembling similar look window angles and applying a propagation model to estimate transmission loss, a partial lobing pattern for the ship’s radiated noise signature can be estimated. A 48-element bottom-mounted hydrophone array in the Canadian Arctic provides an opportunity to test this approach in a low ambient noise environment with a variety of passing ships.

4:10

4pUW11. A comparison of quiet ship certifications with the Enhancing Cetacean Habitat Observation (ECHO) ship noise database. David E. Hannay (JASCO Appl. Sci., 2305-4464 Markham St., Victoria, BC V8Z 7X8, Canada, David.Hannay@jasco.com), Heloise Frouin-Mouy, Jennifer L. Wladichuk (JASCO Appl. Sci., Victoria, BC, Canada), Federica Pace (JASCO Appl. Sci., Eschborn, Germany), and Alexander O. MacGillivray (JASCO Appl. Sci., Victoria, BC, Canada)

Measurements from the large ship noise database acquired by the Vancouver Fraser Port Authority’s Enhancing Cetacean Habitat Observation (ECHO) program were used to assess the conservativeness of five vessel noise certification societies. A multi-variate linear regression analysis of the database was used to scale ECHO measurements to a common reference vessel type for each of 6 categories: tug, tanker, bulker, container ship, vehicle carrier, cruise ship. The purpose of scaling the ECHO measurements was to create a modified dataset to compare with existing vessel noise certification society noise thresholds. The conservativeness of the certification society thresholds was found to vary with vessel category. The general findings are that the society limits are conservative for faster categories (e.g., container ship) but not for slower vessels such as tankers, and certification systems using monopole source level (MSL) had better matches with measurement data than the approaches using radiated noise level (RNL). None of the certification societies accounts for differences of vessels within a vessel category. Therefore, small ships are currently evaluated against the same threshold criteria as large ships. The scaling system developed here using the ECHO dataset could be used to scale measurements (or thresholds) to account for different vessel sizes and operating conditions. This research was funded by Transport Canada.

4:25

4pUW12. Ship source levels in a National Marine Sanctuary. Vanessa M. ZoBell (Biological Oceanogr., Scripps Inst. of Oceanogr., 3869 Miramar St., 2107, San Diego, CA 92092, vmzobell@ucsd.edu), Kaitlin E. Frasier, Sean M. Wiggins, Bruce Thayre, Leila Hatch (National Oceanic and Atmospheric Administration, Scituate, MA), Sean Hastings, Lindsey Peavey (National Oceanic and Atmospheric Administration, Santa Barbara, CA), and John Hildebrand (Scripps Inst. of Oceanogr., La Jolla, CA)

Ship noise in the ocean significantly contributes to low-frequency ambient noise. The Santa Barbara Channel is an ideal site to study noise from shipping because of the intense commercial vessel traffic travelling to the Port of Los Angeles and the Port of Hueneme. In response to concerns of increased ship noise in this area, the Channel Islands National Marine Sanctuary (CINMS) introduced a ship speed reduction program, in the hopes of reducing ship noise. In order to quantify source levels (SLs) of ships that participated in speed reduction, we compared source levels of 31 vessels that participated in the program with source levels of non-participating vessels over the same period. Recordings were made using a High-frequency Acoustic Recording Package at a depth of 580 m, approximately 3 km from the northbound shipping lane. For each ship passage, we measured received levels of passing ships in 1 minute intervals. Source levels (SLs) were calculated by accounting for transmission loss between the recording device and the ship locations obtained from Automated Identification System data. SLs quantified in this work will allow CINMS to evaluate the effectiveness of the speed reduction program for lowering noise inputs from vessels transiting through the sanctuary.

4:40

4pUW13. Small-boat source level estimates in a fjord environment. Ragnhild Smistad (Norwegian Defence Res. Est. (FFI), FFI, Kjeller 2007, Norway, ragnhild.smistad@vegvesen.no), Helge Buen, and Dag Tollefsen (Norwegian Defence Res. Est. (FFI), Horten, Norway)

Small vessels contribute to noise in the marine environment and it is of interest to measure radiated noise levels of such. This paper presents measurements of radiated noise from small leisure boats (lengths 16 to 23 ft) conducted at a shallow water site in a fjord environment. The measurements used a bottom-deployed tripod with three hydrophones, and a laser

instrument for boat range and speed estimation. Data were processed for source levels in one-third octave frequency bands from 0.1 to 1.6 kHz and using the ANSI/ASA standard for distance normalization. For boats moving at speeds 4 to 6 km, source levels varied by up to 20 dB between boat types. Estimated broadband source levels were 8 to 22 dB lower than for merchant vessels previously measured in the same environment.

4:55

4pUW14. Passive acoustic monitoring of ship sounds in Lake Superior. Rosalyn Putland (Dept. of Biology, Univ. of Minnesota Duluth, 1035 Kirby Dr., Duluth, MN 55812, rputland@d.umn.edu) and Allen F. Mensinger (Dept. of Biology, Univ. of Minnesota Duluth, Duluth, MN)

Monitoring freshwater ecosystems using passive acoustics is a largely unexplored approach, despite having the potential to yield information about the biological, geological and anthropogenic activity of a lake or river system. Minnesota, nicknamed "land of 10 000 lakes," provides an interesting case study, because of the opportunity to compare the soundscape during winter with up to 100% ice cover and no vessel transits, to the open water soundscape during busy summer shipping season. Passive acoustic monitoring was conducted in the coastal waters of the western arm of Lake Superior, close the Duluth port, from October 2018 to September 2019. By combining long-term acoustic monitoring data with AIS vessel-tracking data and acoustic propagation modelling (bathymetry, sound speed profiles and seafloor properties) a quantitative method for determining the impact of vessel noise on the soundscape has been established. Median broadband sound pressure level (100–12 000 Hz) was significantly lower in winter compared to summer. However, during ice free months tankers and cargo vessels played a key role in the daily soundscape increasing the intensity by >30 dB between 300 and 1,000 Hz. Moving forward, baseline sound levels provide vital evidence for scientists and governing bodies to make proactive decisions for soundscape conservation.

5:10

4pUW15. Analysis of systematic source level measurements of small vessels. Jennifer L. Wladichuk (JASCO Appl. Sci., 2305-4464 Markham St., Victoria, BC V8Z 7X8, Canada, jennifer.wladichuk@jasco.com), David E. Hannay, Alexander MacGillivray, Zizheng Li (JASCO Appl. Sci., Victoria, BC, Canada), and Sheila Thornton (Fisheries and Oceans Canada, Vancouver, BC, Canada)

Improving our knowledge of how sound impacts marine mammals is particularly important where the spatial distributions of vessels and marine mammals overlap, as exemplified by the critical habitat for the endangered Southern Resident Killer Whale (SRKW). In this study, two acoustic recorders were deployed in transboundary Haro Strait (British Columbia, Canada and Washington State, USA) from July to October 2017 to measure sound levels produced by whale-watching vessels and other small boats. During this period, 20 different volunteer vessels were assessed operating at a range of speeds—nominally 5 knots, 9 knots, and cruising speed. The measurement protocol was designed based on ANSI S12.64-2009. For all vessels, we observed positive correlations between source levels and speed; however, the speed trends (slope of curves) were not as strong as those of large commercial vessels. Mean source levels were computed for each vessel type in the broadband frequency range (0.05–64 kHz), the SRKW communication band (0.5–15 kHz), and the SRKW echolocation band (15–64 kHz) at each speed. Here we discuss how source levels were affected by vessel speed, hull shape and propeller type, as well as the positive and negative aspects of the protocol design.

4p THU. PM

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday, Wednesday, and Thursday evenings. All meetings will begin at 7:30 p.m., except for Engineering Acoustics (4:45 p.m.) and Computational Acoustics (4:30 p.m.).

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 are as follows:

Engineering Acoustics (4:45 p.m.)	Stuart
Acoustical Oceanography	Empress
Animal Bioacoustics	Edison
Architectural Acoustics	Windsor
Physical Acoustics	Wilder
Psychological and Physiological Acoustics	Garden
Structural Acoustics and Vibration	Spreckels

Committees meeting on Wednesday are as follows:

Biomedical Acoustics	Hanover
Signal Processing in Acoustics	Empress

Committees meeting on Thursday are as follows:

Computational Acoustics (4:30 p.m.)	Stuart
Musical Acoustics	Coronet
Noise	Crystal/Continental
Speech Communication	Regent
Underwater Acoustics	Viceroy

Session 5aAA

Architectural Acoustics, Psychological and Physiological Acoustics, Speech Communication, and ASA Committee on Standards: How Does Speech Perception Work: A Tutorial and Panel Discussion for Architectural Speech Privacy

Jennifer Lentz, Cochair
Speech and Hearing Sciences, Indiana University, 200 S. Jordan Ave., Bloomington, Indiana 47405

Kenneth W. Good, Cochair
Armstrong, 2500 Columbia Ave., Lancaster, Pennsylvania

Chair's Introduction—9:00

Invited Papers

9:05

5aAA1. "How": The connection between acoustics and perception. Christian Stilp (Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Speech is both the most important and the most frequent sound that we hear. We have so much experience perceiving speech that it often seems relatively effortless. However, when one examines the speech signal a bit more closely, several severe challenges are evident such as considerable acoustic complexity and extreme acoustic variability across sounds, utterances, talkers, and environments. Given these challenges, how does speech perception work? In this talk, we will consider how speech sounds are produced and observe some of their prevailing acoustic characteristics in American English. We will then embark on a brief tour of the auditory system. We will see how speech sounds are transduced into action potentials in the cochlea and then how the neural representation of speech evolves as one travels up the central auditory system toward the brain. This tour will conclude with the discussion of speech processing in the brain, including cortical pathways for processing different aspects of the speech signal. Understanding speech acoustics and processing can directly inform efforts to provide speech privacy.

9:25

5aAA2. "What" perception of sounds and words. Melissa M. Baese-Berk (Dept. of Linguist, Univ. of Oregon, 1290 University of Oregon, Eugene, OR 97403, mbaesebe@uoregon.edu)

Speech perception is a task that, on the surface, is deceptively simple. Every day in most of our interactions, we are able to understand speech from a variety of talkers about a variety of topics without significant challenges. However, the actual task of understanding speech is a complex one, requiring a listener to quickly extract information from a noisy and variable acoustic signal to determine the speaker's intended meaning. Determining what linguistic features and words a speaker has intended is not trivial, as many factors can influence the acoustic signal. That is, the same intended word or speech segment can be produced with many distinct acoustic realizations depending on the speaker, the surrounding speech context, and other linguistic features, including the speaking rate. Further, speech perception can be even more challenging when a listener is trying to perceive speech in noisy environments, including environmental noise (e.g., construction noise) or competing speech from other talkers. In this talk, I will focus on how listeners are able to extract relatively invariant meaning representations from a highly variable signal. Understanding what is perceived when listeners hear speech is critically important for an understanding of what factors matter for speech privacy.

9:45

5aAA3. "Who": Perception of talker properties. Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, tbent@indiana.edu)

Spoken language simultaneously provides listeners with linguistic content and information about the person speaking. Listeners can identify specific individuals from auditory-only speech signals. Furthermore, listeners can identify a range of physical and socio-cultural talker characteristics (indexical characteristics) with above chance accuracy. Listeners are sensitive to relatively stable aspects of a talker's identity, such as their age, height, gender, race/ethnicity, sexual orientation, region of origin, native, and language status. Talkers also transmit more transitory information (e.g., current health status or emotional state). The mental representations that listeners build for individual talkers and speaker groups interact with the processing of linguistic information, such that the same physical acoustic signal can be interpreted in different ways depending on the talker's perceived indexical characteristics. In this talk, I will focus on how

5a FRI. AM

these mental models can allow for more efficient and accurate processing of speech signals. For example, listeners are more accurate at identifying words from familiar than unfamiliar talkers. Likewise, listeners more accurately understand speech from non-native speakers after a period of exposure. Research on the "who" of speech perception demonstrates the need for considering the interaction between talker characteristics and listener knowledge when making decisions about speech privacy.

10:05–11:05
Panel Discussion

FRIDAY MORNING, 6 DECEMBER 2019 EMPRESS, 8:30 A.M. TO 11:40 A.M.

Session 5aAO

Acoustical Oceanography and Underwater Acoustics: Marine Seismoacoustics

Warren Wood, Cochair
Geology and Geophysics, Naval Research Laboratory, 1005 Balch Blvd, Stennis Space Ctr, Mississippi 39529

Ralph A. Stephen, Cochair
Geol & Geophys MS 24, WHOI, 360 Woods Hole Rd., Woods Hole, Massachusetts 02574

Invited Papers

8:30

5aAO1. Formalizing noise field beam correlation techniques for analyzing deep layers. John Gebbie (Adv. Mathematics Applications, Metron, Inc., 2020 SW 4th Ave., Ste. 170, Portland, OR 97201, gebbie@metsci.com) and Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., Portland, OR)

This work takes an information theoretic approach to understanding and characterizing beam correlations within the ambient noise field and discusses the implications for passive seabed geoacoustic estimation, particularly as it applies to deep layers. The passive fathometer [*J. Acoust. Soc. Am.* **120**(3), 1315–1323 (2006)] and head wave correlation analysis [*J. Acoust. Soc. Am.* **140**(1), EL62–EL66] are examples of techniques that use beam correlations and have generated interest given their ability to expose information about geoacoustic properties as well as the layering structure of the bottom. However, the conditions under which these techniques can operate, what information is actually available, and which multilayer properties can be independently estimated, are open questions. Formal answers come in the form of the Cramér-Rao lower bounds (CRLB), but this requires an exact definition of the probability function for a cross-beamformer output defined in the space of the multilayered bottom properties. Often prior information about a site is available, and this can readily be incorporated to yield the Bayesian CRLB, which allows this approach to be used for experimental design. The theoretical framework for the Bayesian CRLB will be presented along with simulation and measured data examples.

8:50

5aAO2. Seismoacoustic observations at deep ocean observatories: ACO, H2O, and OSN1. Rhett Butler (SOEST, Univ. of Hawaii at Manoa, 1680 East-West Rd., POST 602, Honolulu, HI 96822, rgb@hawaii.edu)

Deep seafloor (~5000 m) observations of seismoacoustic arrivals are presented from the Aloha Cabled Observatory (ACO), Hawaii-2 Observatory (H20), and Ocean Seismic Network-1 (OSN1). Acoustic observations of local earthquakes from the ACO 24 kHz hydrophone show extraordinarily high frequencies up to 165 Hz at distances to ~200 km—a unique window into earthquake source dynamics. Combined seismic and hydrophone observations show that the traditional *T* wave propagates as a seismoacoustic polarized interface wave "*Ti*" coupled to the seafloor. Seismoacoustic *Ti* waves propagating at the sound speed of water are routinely observed over megameter distances at H2O between Hawaii and California, even though the seafloor site is within the shadow zone for acoustic wave propagation. Internal waves or thermohaline covariation may or may not be sufficient to scatter acoustic energy from the SOFAR channel to the deep seafloor. *Ti* has also been observed on seismometers SSW of Oahu at the OSN1 site at the seafloor and within an ODP borehole 242 m into the basalt basement. The observation of Ti from an earthquake in Guatemala at OSN1, whose path is blocked by the Island of Hawaii, is consistent with scattering from the vicinity of the Cross Seamount.

9:10

5aAO3. Stormquakes: Tracing the path of ocean storms through the solid earth. 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), Wenyuan Fan (Dept. of Earth Ocean and Atmospheric Sci., Florida State Univ., Tallahassee, FL), Jeffrey McGuire (U.S. Geological Survey, Menlo Park, CA), Coats Sloan (Woods Hole Oceanographic Inst., Woods Hole, MA), and Julia Fiedler (Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA)

We have discovered a new geophysical phenomenon involving the coupling of the atmosphere-ocean and solid Earth using a novel array method. An analysis of ten years of seismic data recorded in the continental United States, mainly at the USArray, shows that large storms such as hurricanes and Nor'easters can excite transient seismic surface waves in the 20–50 s band with amplitudes equivalent to those excited by M3.5 earthquakes. These sources, which we label stormquakes, can produce transcontinental coherent Rayleigh wave packets observable in the time domain and are thus fundamentally different from previously reported atmosphere-ocean solid Earth couplings that produce incoherent seismic noise. We present unique observations and methods to pinpoint the source locations and timing, which clearly shows that these seismic waves are from interactions of seafloor bathymetry and ocean waves that are energized by large storms. Stormquakes migrate along continental shelfbreak, tracking the leading edge of large storms. We have documented features and identified a possible physical mechanism explaining the stormquake excitation. Stormquakes have potential use in oceanography and meteorology as a remote monitoring tool with high spatial and temporal resolution with which to investigate ocean wave dynamics during large storms.

9:30

5aAO4. EarthScope-Oceans: An array of floating MERMAID instruments for earthquake seismology. Frederik J. Simons (Dept. of GeoSci., Princeton Univ., Guyot Hall 321b, Princeton, NJ 08544-1003, fjsimons@alum.mit.edu), Joel D. Simon (GeoSci., Princeton Univ., Princeton, NJ), Yann Hello (GeoAzur, Sophia-Antipolis, France), Guust Nolet (GeoSci., Princeton Univ., Sophia-Antipolis, France), Masayuki Obayashi (Deep Earth Structure and Dynam. Res., JAMSTEC, Yokosuka, Kanagawa, Japan), and Yongshun J. Chen (Ocean Sci. and Eng., SUSTECH, Shenzhen, Guangdong, China)

Mapping the Earth's uncharted interior through global seismic tomography is dependent on increasing the number of seismic stations in the oceans. We have developed a low-cost, autonomously floating hydrophone to capture earthquake signals suitable for the study of the interior of the Earth and the tectonically and magmatically active underwater realm, while it maintains its potential to be an environmental monitoring device. MERMAID is a freely drifting diver that combines (1) a hydrophone, (2) GPS, (3) an on-board digitizing and processing unit that uses wavelet detection and discrimination algorithms, and (4) an iridium unit for near real-time data transfer with two-way communication. The instrument, 50 kg in air, submersible to 3000 m water depth, with a projected lifetime of up to 5 years, is commercially available from OSEAN SAS. Some 60 units will have been deployed in the Pacific Ocean by the Fall of 2019. With up to 7 kg of sensor payload, additional configurable sensors today include a Sea-Bird SBE 41 CTD, and, in the future, a suite of other instruments with utility in bioacoustics, environmental monitoring, meteorology, bathymetric determination, and chemical and physical oceanography. We report on the performance of our instruments as regards teleseismic event recovery, and on the development of a new method for extracting high-resolution travel times from the first ~1500 seismograms reported live from the Pacific Ocean.

9:50

5aAO5. Mid-ocean ridge eruption rates, long-term climate change, and the importance of marine seismoacoustics. Maya Tolstoy (Columbia Univ., 208 Low Library, New York, NY 10027, mt290@columbia.edu) and Yen Joe Tan (Columbia Univ., Palisades, NY)

Recent work on mid-ocean ridges suggests that eruption rates may not be as steady-state as once thought. There is evidence that seafloor eruptive activity waxes and wanes with long-term climate cycles potentially both being driven by climatic forcings and possibly serving itself as a climatic forcing. An important piece of evidence in understanding mid-ocean ridge volcanic cycles is to look at present-day eruption rates and compare those with expected eruption rates given well-constrained plate spreading rates. However, given that mid-ocean ridge eruptions appear to be relatively infrequent, seismoacoustic coverage of as much of the ocean as possible is key. Work at Axial Seamount and the East Pacific Rise has documented an impulsive signal that is associated with lava erupting onto the seafloor. Thus with adequate marine seismoacoustic coverage, seafloor eruptions can be identified with some confidence without *in situ* confirmation. Such monitoring would provide for better quantification of present-day eruption rates in the marine environment and help understand the contribution of mid-ocean ridges to present-day volcanism, as well as possible long-term variability.

10:10–10:25 Break

Contributed Papers

10:25

5aAO6. Investigation of a rupture-induced underwater sound source. Andrew R. McNeese (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, mcneese@arlut.utexas.edu), Kevin M. Lee, Michael Lee, Jason D. Sagers (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Underwater acoustic experiments, surveys, and sonar often require a repeatable, predictable, and broadband sound source. In many instances, a high amplitude, impulsive sound source is utilized to produce a broadband event capable of penetrating the seabed and propagating to long range. Prior experimentation at the Applied Research Laboratories at the University of Texas at Austin (ARL:UT) has demonstrated the viability of a device that utilizes a rupture disk as an underwater acoustic source. A rupture disk is an expendable diaphragm used in industrial applications, which is designed to break at a specified pressure differential. Placing a rupture disk over an evacuated chamber and mechanically breaking the disk (either by striking on demand or via hydrostatic pressure) at a specified depth was demonstrated to produce high-amplitude, broadband waveforms as the cavity collapses and inflowing water impacts the bottom of the chamber. Residual bubble oscillations are greatly reduced due to the fact that the collapse chamber is initially evacuated. This source also has the advantage that it is solely comprises inert materials—no explosives or combustive gases are

required. Discussion will focus on new and expanded source level measurements and preliminary comparisons to model predictions. [Work Supported by ARL:UT IR&D Program.]

10:40

5aAO7. Observations of compressional, shear and interface waves in the New England Mudpatch. James H. Miller (Ocean Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett Bay Campus URI, Narragansett, RI 02882, miller@uri.edu), Gopu R. Potty (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), Jie Yang (APL-UW, Univ. of Washington, Seattle, WA), and Jason D. Chaytor (U.S. Geological Survey, Woods Hole, MA)

In March 2017, instruments including a tetrahedral hydrophone array and vertically gimbaled geophones were deployed during the Seabed Characterization Experiment in the New England Mudpatch south of Martha's Vineyard. The water depth at the location was about 70 m and a 6-m-thick layer of fine-grained sediments overlays sand layers. The relative positions of the geophones were localized using a time-difference-of-arrival (TDOA) technique applied to signals from Mk-64 SUS charges at various aspects and ranges from 5 to 25 km. In addition, noise from a research vessel provided information on relative depths of the geophones. The geophones were found to have sunk into the soft mud, and this provided a unique opportunity to sense shear waves in the mud and Stoneley waves on the mud-sand interface. In addition, post-Airy Phase arrivals are thought to be shear waves generated at the mud-sand boundary. Interface waves were generated by the interface Wave Sediment Profiler (iWaSP), an instrument with bender beams oriented to shake the seabed. Estimates using iWaSP of shear speed in the mud and sand layers were consistent with inversions using modal dispersion measured on the tetrahedral hydrophone array reported by Potty and Miller (2019). [Work supported by the Office of Naval Research.]

10:55

5aAO8. Environmental characterization using airgun seismic sources. John Hildebrand (Scripps Inst. of Oceanogr., Univ. of California San Diego, Mail Code 0205, La Jolla, CA 92093, jhildebrand@ucsd.edu) and Bruce D. Cornuelle (Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Seismic surveys using arrays of airguns are a ubiquitous feature of the ocean ambient sound environment. An approach is presented to use airguns as a sound source to study how physical oceanographic variability shapes long range acoustic propagation. Signal processing and data analysis tools are presented to allow opportunistic use of commercial seismic surveys in the Gulf of Mexico (GOM) for environmental characterization. Airgun propagation data are analyzed and compared to models generated from conventional physical oceanographic data collection for large scale features such as the Loop Current.

11:10

5aAO9. On the possible role of gravity waves in the ocean in T-phase excitation by earthquakes. Oleg A. Godin (Phys. Dept., Naval Postgrad. School, 833 Dyer Rd., Bldg 232, Monterey, CA 93943-5216, oagodin@nps.edu)

Recent applications of a time-domain spectral-element method to obtain full-wave solutions of seismo-acoustic problems by Bottero *et al.* [*J. Acoust. Soc. Am.*, **144**, EL222–EL228 (2018)] and especially their investigation of T-phases in the Mediterranean [Bottero *et al.*, *J. Acoust. Soc. Am.*, **141**, 4045 (2017)] challenge the existing understanding of the mechanisms of seismic energy conversion into *T*-waves, which propagate at the speed of sound in water. Motivated by these results, this paper aims to evaluate the contribution of scattering by hydrodynamic waves into T-phase generation. Ocean is modeled as a range-independent waveguide with superimposed volume inhomogeneities due to internal gravity waves and surface roughness due to wind waves and sea swell. Guided acoustic waves are excited by volume and surface scattering of ballistic body waves. The size of the effective source of T-waves and their amplitudes are calculated in the single scattering—multiple reflections approximation. Efficiency of the gravity wave-mediated conversion of seismic energy will be compared to that of the conversion that occurs on a seafloor slope.

11:25

5aAO10. Diffraction and scattering at the seafloor. Ralph A. Stephen (RASCON Assoc. LLC, PO Box 567, West Falmouth, MA 02574, rasconassoc@aol.com)

In bottom-interacting ocean acoustics discrete, deterministic arrivals with significant energy are returned from the seafloor at angles that are not predicted by Snell's law. This observation has led to some confusion between the terms "diffraction" and "scattering" as used in ocean acoustics. The common dictionary definitions of diffraction, as the process by which a beam is spread out or bent after passing the edge of an obstacle, and of scattering, as deflecting waves in a more or less random fashion, do not describe the observed seafloor interaction where a pulse of energy is returned deterministically, not randomly, from the seafloor in a non-Snell's law direction. In ocean acoustics and marine seismology, it is rare to consider "bending diffraction" but it is quite common to see "scattering diffraction," for example, as diffraction hyperbolas on bottom profiling and seismic records. For impulsive sources, in addition to specular reflections and random scattering from the seafloor, a significant aspect of bottom-interacting acoustics in deep water is diffraction of energy from discrete seafloor locations along repeatable paths in three-dimensions. [Work supported by ONR.]

Session 5aBA

Biomedical Acoustics: Ultrasound Phantom Development and Tissue Characterization

Yunbo Liu, Cochair
FDA, 10903 New Hampshire Ave., WO62RM2126, Silver Spring, Maryland 20993

Matthew Myers, Cochair
FDA, 10903 New Hampshire Ave., WO62RM2126, Silver Spring, Maryland 20993

Invited Papers

7:30

5aBA1. Characterization of HIFU fields inside of a tissue-mimicking HIFU phantom. Sam Howard (Onda Corp., 1290 Hammerwood Ave., Sunnyvale, CA 94089, sh@ondacorp.com) and Claudio I. Zanelli (Onda Corp., Sunnyvale, CA)

The use of a tissue-mimicking, scatterer-free polyacrylamide gel phantom for HIFU has been previously described [1]. This transparent material combines with a marker with an absorption of 0.6 dB/cm MHz which turns opaque when the temperature exceeds 65 °C. After a brief review of the material properties and their validation, we discuss typical uses of the phantom as part of QA and protocol development and discuss comparisons between phantom and free-field measurements in water at high intensities. Comparisons are also made to new investigations where *in situ* pressures have been measured by embedding a fiber-optic probe into a sample of a phantom material. [1] S. Howard, J. Yuen, P. Wegner, and C. Zanelli, "Characterization and FEA simulation for a HIFU phantom material," in IEEE Ultrasonics Symposium (2003), pp. 1270–1273.

7:50

5aBA2. Holographic measurement and simulation of 3-D ultrasound fields distorted by soft-tissue phantoms. Wayne Kreider (CIMU, Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, wkreider@uw.edu), Alex T. Peek, Christopher Hunter (CIMU, Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Tatiana D. Khokhlova (Dept. of Gastroenterology, Univ. of Washington, Seattle, WA), Pavel B. Rosnitskiy (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Vera A. Khokhlova (Phys. Faculty, Moscow State Univ., Seattle, WA), Petr V. Yuldashev, and Oleg Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

Applications of high intensity therapeutic ultrasound (HITU) rely on the accurate delivery of ultrasound to targeted tissues. However, because *in situ* measurement of 3-D ultrasound fields in tissue is not possible, knowledge of such fields depends upon some form of calculation. Standard approaches utilize measurements in water to characterize the ultrasound source, with fields in tissue estimated by derating or numerical simulation. To evaluate the accuracy of numerical models, simulations and measurements have been performed based on tissue phantoms with known geometries and physical properties. In this work, holography measurements were used to characterize a 256-element HITU transducer in water and to measure directly the 3-D fields behind soft-tissue phantoms designed to introduce refraction, attenuation, and/or aberration. Using synchronization between a waveform digitizer and a computer-controlled positioner, accelerated recording of each hologram (representing over 40 000 points) was performed in about an hour by continuously scanning a capsule hydrophone line-by-line in the measurement plane. Linear measurements are compared with simulations performed using the k-Wave toolbox, and various uncertainties are quantified. It is shown that holography-based modeling accurately represents fields distorted by inhomogeneous layers mimicking body wall. [Funding support by NIH R01-EB025187, R01-EB007643, R01-R01GM122859, and RSF Grant No. 19-12-00148.]

8:10

5aBA3. Models mimicking characteristics of the urinary system and stones relevant to lithotripsy. Adam D. Maxwell (Urology, Univ. of Washington School of Medicine, 1013 NE 40th St., Seattle, WA 98105, amax38@u.washington.edu), Christopher Hunter (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Tony T. Chen (Urology, Univ. of Washington School of Medicine, Seattle, WA), Yak-Nam Wang, Elizabeth Lynch, Barbrina Dunmire, Michael R. Bailey (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), James C. Williams (Anatomy and Cell Biology, Indiana Univ. Purdue Univ. at Indianapolis, Indianapolis, IN), and Wayne Kreider (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Both new and established modes of noninvasive lithotripsy are often tested in an *in vitro* apparatus to evaluate the efficacy of different technologies and exposures to fragment urinary stones. A significant challenge for these experiments is to accurately replicate the physical properties of a stone and the surrounding urinary tract. Such a task requires appropriate models for the stone, the fluid, and the tissue, all of which can contribute to stone fragmentation. This presentation will describe research into new models to more accurately replicate the fracture of natural human stones, including artificial models and natural stones from other mammalian species. Experiments show several factors such as fluid gas concentration, tissue phantom properties, and fluid confinement and also affect cavitation behavior and lithotripsy effectiveness. Experiments measuring such cavitation activity *in vivo* in a porcine model and correlation with *in vitro* observations will be presented. Altogether, these studies suggest that lithotripsy experiments are dependent on these characteristics and require careful choice of models to mimic *in vivo* scenarios. [Work supported by NIDDK K01 DK104854 and P01 DK043881.]

Contributed Papers

8:30

5aBA4. Simulation of acoustic backscattered field from a medium with randomly distributed fibers. Mohammadreza Kari (Medical Phys., UW Madison, 703 Eagle Heights, Apt. I, Madison, WI 53705, mkari@wisc.edu), Helen Feltovich (Maternal Fetal Medicine, Intermountain Healthcare, Provo, UT), and Timothy J. Hall (Medical Phys., UW Madison, Madison, WI)

Common assumptions in quantitative ultrasound include that the medium is homogeneous and isotropic and has diffuse scattering conditions. In some soft tissues, due to the presence of fiber-like structures, the medium is anisotropic and has a periodic scattering condition. Skeletal muscle is an example in which the fibers are highly aligned, a coherent scattering condition appears. Using the closed form solution of acoustic backscattered field from immersed finite cylindrical targets, this work presents a simulation study of a medium containing randomly distributed fibers. The effective scatterer diameter and average backscatter coefficient of the medium are calculated and compared with the change in fiber orientation. The effects of other parameters such as number of fibers and their lengths are also analyzed. In addition, different distributions of parameters such as normal or uniform distributions are considered, and their effects on the backscattered field are studied.

8:45

5aBA5. Experimental investigation of acoustic backscattered field from a medium with random and aligned fibers. Mohammadreza Kari (Medical Phys., UW Madison, 703 Eagle Heights, Apt. I, Madison, WI 53705, mkari@wisc.edu), Helen Feltovich (Maternal Fetal Medicine, Intermountain Healthcare, Provo, UT), and Timothy J. Hall (Medical Phys., UW Madison, Madison, WI)

Some soft tissues, such as skeletal muscles, have fiber-like structures that make them anisotropic. Acoustic backscattering from such tissues or media has been of interest to many scientists in biomedical fields because it complicates interpretation of imaging data. Backscatter coefficient measurements show diagnostic promise for this purpose both *in vitro* and *in vivo*. Many parameters, e.g., effective scatterer diameter or average scattering strength, can be estimated with acoustic backscatter measurements. In the present work, we experimentally study the backscatter coefficient of a medium with aligned or random fibers and estimate the effect of fiber orientation upon the backscattered field and the estimated scatterer diameter. Using an experimental approach, we explore the relationship between the backscatter coefficient of a medium with multiple fibers and with a single fiber. In addition, we investigate the effect of the spatial Fourier transform of the fiber distribution in the medium with its frequency dependence of backscatter coefficients. The experimental results show that backscatter coefficients of a medium with random fibers can be modeled by the product of number density of fibers, backscatter coefficients of the same single fiber, and the spatial Fourier transform of the fiber distribution.

9:00

5aBA6. Aberration correction using nonlinear backscattered signals from the focus of an ultrasound beam. Christopher R. Bawiec (Div. of Gastroenterology, School of Medicine, Univ. of Washington, 325 9th Ave., Harborview Medical Ctr., Box 359634, Seattle, WA 98103, bawiec@uw.edu), Vera A. Khokhlova (CIMU, APL, Univ. of Washington/Phys. Faculty, Moscow State Univ., Seattle, WA), Oleg A. Sapozhnikov (CIMU, APL, Univ. of Washington/Phys. Faculty, Moscow State Univ., Moscow, Russian Federation), Matthew O'Donnell (Dept. of BioEng., Univ. of Washington, Seattle, WA), Christopher Hunter, Mohamed A. Ghanem, Alex T. Peek (CIMU, APL, Univ. of Washington, Seattle, WA), and Tatiana D. Khokhlova (School of Medicine, Div. of Gastroenterology, Univ. of Washington, Seattle, WA)

High intensity focused ultrasound (HIFU) therapies are often affected by aberrations, which can severely limit treatments. For multi-element arrays, applying phase corrections for each element could mitigate these effects. Here, an aberration correction method based on cross-correlation of nonlinear HIFU waves, backscattered from the focus is proposed and tested. A spherically focused 1.5 MHz 256-element HIFU array powered by Verasonics system was used for both emitting and receiving backscattered ultrasound waves. A short (3-cycle) high-amplitude pulse was transmitted through an aberrating layer into an *ex vivo* clotted bovine blood sample. The backscattered signal was recorded by individual array elements and high-pass filtered at 4 MHz. The layer introduced different acoustic path lengths for individual elements leading to a decrease in the peak-positive pressure of ~70%. The time delay required for each element to restore the nonlinear waveform at the focus was calculated based on the cross-correlation between the backscattered filtered signals. The results were confirmed by replacing the sample with a hydrophone. Using the corrected time delays, peak-positive focal pressure increased over 75% from uncorrected aberrated levels. These results demonstrate that the proposed focal "nonlinear beacon" can be used for phase correction feedback in HIFU treatments of soft tissue. [Work supported by NIH R01GM122859, R01EB007643, and RSF19-12-00148.]

9:15

5aBA7. Quantification of lung surface wave speed in a water-filled lung: An *ex vivo* swine lung study. Xiaoming Zhang (Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, zhang.xiaoming@mayo.edu), Boran Zhou, and Alex X. Zhang (Mayo Clinic, Rochester, MN)

Lung edema is a common symptom of congestive heart failure and inflammatory conditions, such as acute respiratory distress syndrome. Currently, computed tomography (CT) is considered the standard to assess lung edema, but it requires ionizing radiation and poses a significant logistic burden. Lung ultrasound (LUS) has previously been used to assess lung edema using ultrasound B-line artifacts, but analysis of the B-line artifacts relies on visual interpretation. We developed lung ultrasound surface wave elastography (LUSWE) to measure lung surface wave speed safely. This project aims to evaluate LUSWE to quantify lung surface wave speed in an *ex vivo* water-filled swine lung. The lung surface wave speeds were measured at baseline and at frequencies of 100 Hz, 200 Hz, 300 Hz, and 400 Hz. Then, an amount of water was filled into the lung through its trachea. Ultrasound imaging was used to guide the water filling until significant changes were visible on the imaging. The lung surface wave speeds were measured. An additional 120 ml of water was then filled into the lung. The lung surface wave speeds were measured again. The results demonstrated that the lung surface wave speed decreased with respect to water content.

9:30

5aBA8. Modelling Lamb waves in the septal wall of the heart. Alberico Sabbadini (Acoust. Wavefield Imaging, ImPhys, Faculty of Appl. Sci., Delft Univ. of Technol., Lorentzweg 1, Delft 2628 CJ, The Netherlands, a.sabbadini@tudelft.nl), Annette Caenen, Hendrik J. Vos (Biomedical Eng., Dept. of Cardiology, Erasmus MC, Univ. Medical Ctr. Rotterdam, Rotterdam, The Netherlands), Nico de Jong, and Martin D. Verweij (Acoust. Wavefield Imaging, ImPhys, Faculty of Appl. Sci., Delft Univ. of Technol., Delft, The Netherlands)

Shear Wave Elastography (SWE) has been proposed to investigate cardiac health by non-invasively monitoring tissue stiffness. Previous work has shown that the plate-like geometry of the Interventricular Septum (IVS) may result in a dispersion similar to Lamb waves, complicating the link between shear wave speed and cardiac stiffness. However, the IVS is not a simple plate, e.g., its thickness tapers across its length. We have used 2-D Finite Element simulations to investigate the effects of tapering on Lamb waves. The model consists of an elastic slab immersed in water, with a thickness decreasing smoothly in space from 9 to 3 mm. Pulses with low (0–80 Hz) and high (0–700 Hz) frequency contents were used to excite natural and acoustic radiation force induced waves. The results show that, at the lower frequencies, propagation speed can decrease during propagation by ~20% due to the thickness reduction, producing a nonlinear space-time relation from which multiple speed values can be extracted. At higher frequencies, the main observation is a dependence of the dispersion behavior on the shape of the tapering (e.g., linear, concave, or convex). These results suggest that septal geometry is likely to play a role in deriving cardiac stiffness from propagation speed measurements.

9:45–10:00 Break

10:00

5aBA9. Ultrasound elastography monitoring reveals a decline in shear wave attenuation and clot viscosity after recombinant tissue plasminogen activator treatment of blood clots. Guillaume Bosio (Univ. of Montreal Hospital Res. Ctr., Universite de Montreal, 900 St. Denis, Montreal, PQ, Canada, guillaume.bosio@umontreal.ca), Manish Bhatt, and Guy Cloutier (University of Montreal Hospital Res. Ctr., Universite de Montreal, Montreal, PQ, Canada)

Deep vein thrombosis is one of the leading causes of disability and serious illness and can become fatal. The lytic recombinant tissue plasminogen activator (rt-PA) is the main drug used for clot lysis and rapid normalization of venous blood flow. Less than 40% of patients who receive rt-PA treatment have improved blood flow. In this study, we quantitatively monitor the rt-PA treatment in *in-vitro* blood clots with ultrasound elastography. An acoustic radiation force imaging sequence was implemented on a research ultrasound system to remotely generate shear waves inside the blood clot samples. Porcine blood samples from two different pigs were bought from a local slaughterhouse. Clots of varying viscoelasticity were prepared by allowing different coagulation time between 30 min to 3 days in borosilicate glass pipettes. The clots were then embedded in gelatin-agar phantoms to perform ultrasound measurements. Another batch of clots from the same blood samples was treated with rt-PA drug for 30 min, and ultrasound measurements were performed after treatment. In 11 samples, variations in SW speed before and after rt-PA treatment were not found to be statistically significant (>0.05). However, SW attenuation and clot viscosity declined significantly after rt-PA treatment by 33.49% ð 31.07% (<0.05) and 33.33% ð 35.72% (<0.05). The results indicate that SW attenuation and viscosity can be used to monitor DVT treatment, and clot viscosity may emerge as an important biomarker for clot staging.

10:15

5aBA10. Shear wave attenuation measurement of tissue-mimicking materials using a two-point frequency shift method. Piotr Kijanka (Dept. of Radiology, Mayo Clinic, 200 First St. SW, Rochester, MN 55905, kijanka.piotr@mayo.edu) and Matthew W. Urban (Dept. of Radiology, Mayo Clinic, Rochester, MN)

Shear wave elastography (SWE) is a method used in several clinical applications for assessment of soft tissue viscoelastic properties. SWE uses an acoustic radiation force (ARF) to produce laterally propagating shear waves to obtain the wave velocity. Here, we present a two-point shear wave attenuation coefficient measurement method without using a rheological model. The technique uses information related to the magnitude spectrum frequency shift of shear waves measured at only two lateral locations. We examined how the distance from the start of the ARF push and the distance between the two locations affected the attenuation coefficient estimates. We tested this method on digital phantom data created using local interaction simulation approach in viscoelastic media and on data acquired from phantom and *ex vivo* liver experiments. We compared the results from the two-point method with other two techniques used for assessing shear wave attenuation: the frequency-shift-based (FS) method and attenuation measuring ultrasound shearwave elastography. In comparison with the FS method, our technique does not assume that the shape parameter of the Gamma function is constant over a lateral distance. Tests conducted showed that the proposed method is feasible to provide robust attenuation estimates based on two measurement points in tissue-mimicking materials and *ex vivo* liver tissue.

10:30

5aBA11. Measurements of ultrasound attenuation in human chest wall. Brandon Patterson (Radiology, Univ. of Michigan, 1301 Catherine St., Ann Arbor, MI 48109, awesome@umich.edu) and Douglas Miller (Radiology, Univ. of Michigan, Ann Arbor, MI)

Knowledge of the bulk acoustic properties of human chest wall is useful in the development and study of cardiac and lung ultrasound exposimetry. However, few studies have been performed on human tissue, and there is a need for attenuation data relevant to clinical diagnostic ultrasound. In this study, we measured the bulk acoustic attenuation in unembalmed, never-frozen human chest wall samples, using a GE Vivid 7, clinical diagnostic ultrasound machine. B-mode ultrasound with frequencies from 1.6 to 5.0 MHz was transmitted through human chest wall tissue samples and compared to equivalent-source signals transmitted through normal saline. The recorded signals were analyzed to characterize the acoustic attenuation. Preliminary results based on 9 donors show that both chest wall morphology and bulk attenuation properties vary widely between individuals. Chest wall thicknesses ranged from 2 to 6 cm. The mean linear acoustic attenuation coefficients, across all samples, ranged from 1.1 dB/cm MHz at 1.6 MHz to 1.7 dB/cm MHz at 5.0 MHz, with a frequency-averaged standard deviation of 0.4 dB/cm MHz. Across all donors and frequency and power settings, linear acoustic attenuation coefficients ranged from 0.5 to 2.3 dB/cm MHz. Variation in thoracic exposure may result from variations in the chest wall composition and morphology between patients.

Invited Paper

10:45

5aBA12. A closer look at ultrasonic attenuation in a tissue-mimicking material. Subha Maruvada (U.S. Food and Drug Administration, 10903 New Hampshire Ave., Bldg. WO 62-2222, Silver Spring, MD 20993, subha.maruvada@fda.hhs.gov), Yunbo Liu, Joshua Soneson, Bruce Herman, and Gerald Harris (U.S. Food and Drug Administration, Las Vegas, NV)

A well-characterized ultrasound tissue-mimicking material (TMM) can be important in determining the acoustic output and temperature rise from high intensity therapeutic ultrasound (HITU) devices and also in validating computer simulation models. A HITU TMM previously developed and characterized in our laboratory has been used in our acoustic and temperature measurements as well as modeled in our HITU simulation program. A discrepancy between the thermal measurement and simulation, though, led us to further investigate the TMM properties. We found that the 2-parameter analytic fit commonly used to represent the attenuation of the TMM in the computer modeling was not adequate over the entire frequency range of interest, 1 MHz to 8 MHz in this study, indicating that we and others may have not been characterizing TMMs, and possibly tissue, optimally. By comparing measurements and simulations, we found that a 3-parameter analytic fit for attenuation gave a more accurate value for attenuation at 1 MHz and 2 MHz. When using a 3-parameter fit, the temperature rise measurements in the FDA TMM agreed more closely with the simulation results.

11:05

5aBA13. Novel bilayer aberration-inducing gel phantom for high-intensity focused ultrasound applications. Alex T. Peek (CIMU Appl. Phys. Lab., Univ. of Washington, 1715 NE Columbia Rd., Seattle, WA 98195, apeek@uw.edu), Tatiana D. Khokhlova (Dept. of Medicine Gastroenterology, Univ. of Washington, Seattle, WA), Pavel B. Rosnitskiy, Petr V. Yuldashev (Acoust., Phys. Faculty, M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), Christopher R. Bawiec (Dept. of Medicine Gastroenterology, Univ. of Washington, Seattle, WA), Wayne Kreider, Christopher Hunter (CIMU Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Oleg Sapozhnikov (Acoust., Phys. Faculty, M.V. Lomonosov Moscow State Univ., Moscow, Russian Federation), and Vera A. Khokhlova (CIMU Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Aberrations induced by soft tissue inhomogeneities often complicate HIFU therapies. In this work, a bilayer phantom made from polyvinyl alcohol hydrogel (PVA) and ballistic gelatin (BG) was built to mimic alternating layers of water-based and lipid tissues characteristic for a body wall and reproducibly distort HIFU fields. The density (1.04 g/ml PVA, 0.86 g/ml BG), sound speed (1520 m/s PVA, 1450 m/s BG), attenuation coefficient (0.016 dB/cm MHz PVA, 0.052 dB/cm MHz BG) of gel materials, and nonlinearity coefficient close to that of water were measured using homogeneous gel layers of 4 cm thickness. A 3-D-printed mold was fabricated to generate the random interface between the gel layers. The interface pattern was designed using a 2-D Fourier spectrum approach replicating different spatial scales of tissue inhomogeneities. Distortion of the field of a 256-element 1.5 MHz HIFU array by the phantom was characterized through hydrophone measurements for both linear and nonlinear beam focusing. Both spatial shift (up to 2.3 mm axially and 0.35 mm transversely) and widening of the focus were observed, as well as dramatic reduction in focal pressures caused by aberrations (up to fourfold decrease in the peak positive pressure for a focal waveform with developed shock). [Work supported by NIH R01EB007643, R01GM122859, and R01EB025187.]

11:20

5aBA14. Direct error in constitutive equation formulation for inverse problems in time harmonic viscoelasticity. Paul E. Barbone (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, barbone@bu.edu) and Olalekan A. Babaniyi (Appl. Mathematics, Rochester Inst. of Technol., Boston, MA)

We present a variational formulation for inverse viscoelasticity problems based on minimizing the error in a constitutive equation. This is formulated as a quadratic minimization problem with linear constraints and, hence, is amenable to a direct solution with a single solve. Here, we extend our prior work with scalar waves to inverse viscoelasticity, intended for application to shear wave fields excited by acoustic radiation force pulses. We demonstrate that the problem is ill-posed with a single wave field but is stable with two measured wave-fields. We compare two approaches using two input wave-fields: first, we consider using two wave fields excited by two different push pulse configurations. Second, we consider using a single push pulse but combining information across frequencies.

11:35

5aBA15. Flow phantom for contrast enhanced ultrasound research, device validation, and clinical training. Thomas Matula (Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, matula@uw.edu), John Kucewicz, and Barbrina Dunmire (Univ. of Washington, Seattle, WA)

Phantoms are an important part of ultrasound research, instrument development, and clinical training. There are, however, no standardized phantoms for contrast enhanced ultrasound. Phantoms typically include either tubes to mimic flow in larger vessels or dialysis tubing to mimic perfusion in the capillary bed. These approaches fail to mimic the full range of flow regimes and tissue scattering observed in the body during imaging. To address some of these limitations, we designed a phantom that integrates flow in both the macro- and microcirculation with physiological tissue backscatter, attenuation, and sound speed and that can be customized to simultaneously mimic flow in normal and malignant tissues. A variety of sponges and foams were considered. Acceptable materials were fit in a custom-built flow chamber and imaged with a commercially available ultrasound system. A polyvinyl alcohol (PVA) sponge was deemed to be a suitable phantom material. PVA has a sound speed of 1490 m/s and attenuation coefficient of 0.6 dB/cm/MHz. After introducing a contrast agent into the flow system, the agent could be visualized rapidly entering the flow inlets, perfusing slowly through the sponge, and rapidly exiting through the flow outlets. [Funded by Philips Ultrasound and the Life Sciences Discovery Fund No. 3292512.]

FRIDAY MORNING, 6 DECEMBER 2019 HANOVER, 8:30 A.M. TO 10:45 A.M.

Session 5aCA

Computational Acoustics: General Topics in Computational Acoustics

Jennifer Cooper, Cochair

Johns Hopkins University Applied Physics Laboratory, 11100 Johns Hopkins Rd., Mailstop 8-220, Laurel, Maryland 20723

Mallory M. Morgan, Cochair

Rennselaer Polytechnic Institute, Greene Bldg, RPI, Troy, New York 12188

Contributed Papers

8:30

5aCA1. Numerical modeling of collinear mixing of compressional and shear waves in nonlinear elastic media using the iterative nonlinear contrast source method. Sundaraelangovan Selvam (Acoust. Wavefield Imaging, Delft Univ. of Technol., Lorentzweg 1, Delft 2628CJ, The Netherlands, s.selvam@tudelft.nl), Arno Volker, Paul L. van Neer (Acoust. and Sonar, TNO, The Hague, The Netherlands), Nico de Jong, and Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., Delft, The Netherlands)

In nondestructive testing, nonlinear wave mixing could be used to obtain the nonlinearity parameters of an elastic medium and thereby get information about its state, e.g., aging and fatigue. To better understand the mixing mechanisms and optimize the design of measurement setups, a physics-oriented tool for the simulation of nonlinear elastic wave propagation would be valuable. In this presentation, we extend the iterative nonlinear contrast source method (INCS) to study the nonlinear mixing of two plane, collinear bulk waves (one compressional, one shear) in a homogeneous, isotropic, elastic medium with two independent coefficients of nonlinearity (β_L and β_T). The method successfully captured the resonant wave generated due to the mixing (one-way and two-way) of primary waves of different frequencies. The obtained results for the resonant wave were in good agreement with the results reported in the literature. In addition, the contrast source allowed the propagating and evanescent components of the scattered wave field to be studied in the wavenumber-frequency domain, which provides physical insight into the mixing process and explains the propagation direction of the scattered wave. Thus, the INCS method seems to be a useful tool to investigate and predict wave mixing in nonlinear elastic media.

8:45

5aCA2. Parallel computing using Python-based software for a high-frequency ultrasound system. Alejandro I. Villalba (Elec. and Comput. Eng., Dalhousie Univ., 5790 University Ave., Rm. 246, Halifax, NS B3H IV7, Canada, alejandro.villalbae@gmail.com), Thomas Landry, and Jeremy Brown (Biomedical Eng., Dalhousie Univ., Halifax, NS, Canada)

Python has become a widely used programming language for scientific research. However, python's computational speed is limited when compared with other languages. This work presents an upgraded open-source imaging software based on python, which is capable of acquiring, processing, and displaying B-mode images in real-time for a high-frequency phased array imaging system. Ultrasound systems, considered to be real-time, display images at a frame rate of >20 Hz. This software uses 3 central processing units (CPUs) for parallel processing. In addition, Graphics Processing Unit (GPU) parallel computing is achieved by employing OpenCL commands and executing them directly on the GPU. New features, such as frame weighted averaging, color flow overlay and image enhancement, were also implemented. Parallel computing increased the displayed frame rate in general B-mode from 20 Hz to 35 Hz. B-mode + color flow overlay was increased from 15 Hz to 29 Hz and enhanced B-mode (applying edge detection, blur, and FFT filtering) was increased from 14 Hz to 28 Hz. Finally, enhanced B-mode + color flow overlay frame rate was increased from 8 Hz to 23 Hz. Thus, even more advanced ultrasound techniques, such as shear wave elastography might be developed using the python language without compromising the frame rate.

9:00

5aCA3. Applying neural networks to computational simulations of ultrasonic scans to determine scatterer location. Nguyen Nguyen, Eleanor Goff, Naomi Brandt (Phys., Mount Holyoke College, South Hadley, MA), and Maria-Teresa Herd (Biological and Physical Sci., Assumption College, 50 College Ave., South Hadley, MA 01075, therd@mtholyoke.edu)

In the field of ultrasound imaging, it has been theorized that speckle may be the product of unresolvable scatterers. These unresolved scatterers may contain information about tissue structure, including the development of small tumors. Unpacking this information from speckle patterns is mathematically untenable, but neural networks can be used to recognize these patterns. As of now, few attempts have been made to apply neural networks to predict scatterer placement from ultrasound scans. We have constructed a computational simulation which consistently and accurately reproduces experimental data and can be used to train a neural network. Using Field II, a Matlab-based program for ultrasound modelling, simulations were created to replicate data produced from experimental phantoms composed of tapioca beads and agarose gel. Simulations were designed to account for bead placement and size, as well as experimental conditions. Comparisons between simulated and experimental data using statistical analysis show that computational methods can accurately predict ultrasound images. This verification allows us to train a neural network using simulated data and apply this training to analyze experimental data.

9:15

5aCA4. Automatic speech emotion recognition using deep learning for analysis of collaborative group meetings. Mallory M. Morgan (Rennselaer Polytechnic Inst., Greene Bldg, RPI, Troy, NY 12180, morgam11@rpi.edu), Indrani Bhattacharya, Rich Radke, and Jonas Braasch (Rennselaer Polytechnic Inst., Troy, NY)

Emotion is a central component of verbal communication between humans. Due to advances in machine learning and the development of affective computing, automatic emotion recognition is both increasingly possible and increasingly sought-after. In this project, deep learning architectures, such as long short-term memory networks, are used to classify emotion from the prosody of human speech. Emotional groupings are considered both categorically (i.e., happiness, anger, etc.) and with respect to a three-dimensional emotion space divided into axes of valence, activation, and dominance. The network is trained on publicly available emotional speech

corpuses and tested on a new dataset as part of a larger project to understand the dynamics of collaborative groupwork. Specifically, small groups were asked to complete a lunar survival task while recorded in a smart meeting room equipped with multimodal sensors. The results of the emotion classification, as well as other non-verbal speech metrics, are correlated with user-reported rankings including emergent group leader and major group contributor, in the hopes of better understanding the complex dynamics of team meetings. [Work supported by NSF IIP-1631674, a Northeastern University Tier 1 Seed Grant, and the Cognitive and Immersive Systems Laboratory (CISL).]

9:30–9:45 Break

9:45

5aCA5. Using graphics programming methods in PC SWAT. Denton Woods (NSWC PCD, 110 Vernon Ave., Panama City, FL, denton.woods@navy.mil)

The Panama City Shallow Water Acoustic Toolset (PC SWAT), developed at the Naval Surface Warfare Center Panama City Division (NSWC PCD), simulates sonar performance for a wide variety of underwater environments. Recent technical enhancements in the simulation package include the addition of triangular target scattering using the Kirchhoff approximation, enabling the use of arbitrary 3-D targets. Technical inspiration and techniques borrowed from the graphics programming community that have been adapted to improve performance predictions and fidelity of the sonar simulator package will be presented.

10:00

5aCA6. Numerical method for prediction of duct break out sound power. Paul T. Williams (Univ. of Technol. Sydney, 32/34 Lord St., Botany, New South Wales 2019, Australia, paul.williams@uts.edu.au), Ray Kirby (Univ. of Technol. Sydney, Ultimo, New South Wales, Australia), and James Hill (AAF Ltd., Cramlington, United Kingdom)

The acoustic design of duct systems requires consideration of both the noise propagating within a duct and also of the noise transmitted out through the duct walls into the environment. Control of this breakout noise can form a significant part of a noise control solution, and so understanding this phenomenon can lead to reduced material use. The breakout noise is investigated here using coupled structural-acoustic finite element models. Propagation along a waveguide with constant cross section is represented using a modal expansion of the acoustic pressure in the fluid and displacement in the duct walls. The free-field external environment requires an outer boundary condition, and for this purpose, a perfectly matched layer is applied at some distance from the elastic walls. The finite length of the waveguide is then enforced by coupling the fields to separate infinite length inlet and outlet ducts by the mode matching method. Transmitted power from the finite length duct is investigated when a noise source is placed in the inlet.

10:15

5aCA7. AI methods for duct acoustics. Stefan Sack (The Marcus Wallenberger Lab., The Royal Inst. of Technol., Teknikringen 8, Stockholm 100 44, Sweden, ssack@kth.se), Cheng Yang (Inst. of Vib. Shock and Noise, School of Mech. Eng., Shanghai Jiao Tong Univ., Shanghai, China), and Mats ûbom (The Marcus Wallenberger Lab., The Royal Inst. of Technol., Stockholm, Sweden)

Acoustic mode decomposition is commonly used to separate pressure wave components in flow ducts, and established methods relay on mathematical descriptions of the wave motion. In this work, a new approach to mode decomposition is presented which uses artificial intelligence (AI) to separate the acoustic mode content. A neural network is trained with data gained from a small set of numeric solutions of the Linearized Navier-Stokes equation in the frequency domain in a straight duct. The network is tested on relevant experimental data. Good agreement with the analytical method is demonstrated in two applications with different flow conditions. We conclude that using AI for mode decomposition is a promising alternative to the standard method as it is applicable to a much broader set of aeroacoustic problems.

10:30

5aCA8. Helmholtz-Kirchhoff integral formula for predicting the acoustic pressure from a radiator including irregular surface. Kyounghun Been, Junsu Lee (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), Pohang-si, South Korea), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), PIRO 405, POSTECH, San31, Hyoja-dong, Nam-gu, Pohang, Kyungbuk 790784, South Korea, wkmoon@postech.ac.kr)

The Helmholtz-Kirchhoff integral (HKI) for the radiation problem from a finite source with a vibrating surface is a method of calculating the acoustic pressure at any position in the three-dimensional space using the velocity conditions of the radiating surface. Previous studies predicted the acoustic pressure from the radiator based on the HKI. However, the acoustic pressure should be evaluated first on the radiation surface in order to calculate the acoustic pressure in the space. Nonetheless, it is difficult to evaluate the acoustic pressure on the radiation surface accurately if there are non-smooth points. Because the exact HKI is not easily found to be applicable for the evaluation, in this study, we investigated the HKI from the beginning. Through the process of deriving the formula, we proposed the form of the HKI that can be used for calculating the acoustic pressure at both smooth and non-smooth points of the radiation surface. The proposed formula is also proved mathematically. As a result, the acoustic pressures calculated by the proposed formula are compared to those using a FEM. The results yield that the overall relative errors are less than 2% even on the radiator surface including vertices. [Work supported by Samsung Display Corporation.]

Session 5aNS

Noise, Physical Acoustics, and Computational Acoustics: Jet Noise Reduction Workshop

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

ASA is partnering with the United States Jet Noise Reduction Science and Technology Panel to hold this workshop targeting emerging trends and technologies in jet noise solutions.

Transformative solutions to reduce the impacts of noise from supersonic jet propelled aircraft, particularly military fixed-wing fighters, are driven by national health and safety requirements to protect the hearing of warfighters and political pressures from communities in areas of high military aircraft activity. Jet noise reduction technology is limited to solutions that do not affect military aircraft performance requirements to fly faster and farther. In this workshop, distinguished leaders will present special lectures regarding the current climate and future outlook for the most promising jet noise reduction solutions. They will also discuss real-world applications in a panel Q&A session.

FRIDAY MORNING, 6 DECEMBER 2019 WILDER, 8:00 A.M. TO 11:15 A.M.

Session 5aPA

Physical Acoustics: General Topics in Physical Acoustics IV

John S. Allen, Chair
Mechanical Engineering, Univ. of Hawaii -Manoa, 2540 Dole Street, Holmes Hall 302, Honolulu, Hawaii 96822

Contributed Papers

8:00

5aPA1. Acoustics of water entry of hydrophobic spheres. Rafsan Rabbi, Tadd Truscott (Mech. & Aerosp. Eng., Utah State Univ., Logan, UT), John S. Allen (Mech. Eng., Univ. of Hawaii -Manoa, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu), and Jesse Belden (Navy Undersea Warfare Ctr., Newport, RI)

Hydrophobic spheres form different cavities depending on the sphere size and the impact parameters. Cavities types have been investigated with respect to their interface characteristic, and regimes denoted as deep seal, surface seal, shallow seal, quasi-static seal, and rebound have been previously reported. The acoustic characteristics of these cavities have relatively unexplored compared with that of impacting liquid drops. A previously developed experimental setup for water entry visualization is extended for acoustic measurements. Simultaneous acoustic and optical measurements are done with the synchronization of a condenser air microphone and hydrophone with two high speed video cameras. Spheres of diameter 8–40 mm are investigated for corresponding impact velocities of 1.4–6.0 m/s. Room temperature metal spheres sprayed with Cytonix WX-2100 coated are compared with those heated up to the inverse Leiden temperature (120–440°C). In both cases, acoustic signature is found to be directly related to sealing phenomena. Upon impact, the heat spheres have a specific vaporization sound compared to the coated case. The transition and values of the Leiden frost temperature are investigated with respect to spectral acoustics signal features.

8:15

5aPA2. Impact of viscoelastic spheres on to hydrophobic and hydrophilic substrates with liquid pools. Rafsan Rabbi, Tadd Truscott (Mech. & Aerosp. Eng., Utah State Univ., Logan, UT), and John S. Allen (Mech. Eng., Univ. of Hawaii-Manoa, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu)

The study of the sound of impact has an extensive history with seminal studies done by Rayleigh. Rayleigh studied the subsequent vibrations of an impacting elastic sphere on a solid surface formulating a preliminary, analytical theory for rigid body radiation based on energy considerations. For a spherical particle incident on a non-wetted substrate, the sound pressure level can be related to the particle's impulsive acceleration. Impact onto wetted substrates is less understood for the cases of hydrophobic surfaces. We investigate the impact of visco-elastic water beads (12.5 mm, 17.5 mm diameter) from heights 35–400 cm upon hydrophilic and hydrophobic liquid pools (0.15–0.45 ml). A hydrophobic surface is obtained by coating the metallic surface of a contact microphone with Soft99 Glaco Mirror Coat Zero

5a FRI. AM

water repellant. Sound production is measured using both contact and air microphones, which are synchronized with a high-speed camera (Phantom V2512) for visualization. In the hydrophobic case at low heights, the rebound of the water bead results in the attachment and lift-off of the liquid. The collapse of entrained gas cavities corresponds to an oscillatory acoustic signature upon lift-off. At higher impact heights, the liquid pool is atomized with signatures of drop break-up.

8:30

5aPA3. Experimental modeling of mass eruption rates from acoustics wave. Claudia SûÀnchez (Dipartimento di Scienze della Terra, Universitû degli Studi di Firenze, Via Giorgio La Pira, 4, Firenze 50121, Italy, pox.sancheza@gmail.com), Maurizio Ripepe, Giorgio Lacanna (Universitû degli Studi di Firenze, Firenze, Italy), and Pascuale Poggi (Istituto nazionale di ottica del Cnr di Firenze, Firenze, Italy)

Among the different volcanoes dynamics encountered worldwide, the repetitive bursting of giant, elongated gas bubbles ("slugs") occupying the whole conduit diameter is a feature commonly described as Strombolian activity. One of the key parameters to understand this eruptive behavior is the estimate of the overpressure ΔP inside the bubble before its explosion, which may occur either at the volcano vent or inside the conduit. This quantification, however, is still a major challenge. On the one hand, it has been shown that for weak acoustic waves (linear regime), the amplitude of the pressure wave propagating in the atmosphere due to the bursting does not depend linearly on ΔP. On the other hand, the gas pressure inside volcanic bubbles may strongly vary and sometimes exceed the atmospheric pressure by up to several MPa, exhibiting non-linear acoustic regimes. We investigate experimentally the acoustic signal released by a cylindrical, overpressurized cavity initially closed. We explore the transition between the linear and non-linear regimes and their relationship with the velocity fluid and mixture particles on the system for estimating the mass flow rate using Lighthill's equation.

8:45

5aPA4. Mesocopic wave physics in a dense fish school. Benoit Tallon (ISTerre, CNRS, 1381 rue de la Piscine, Saint martin d'heres 38400, France, benoit.tallon@univ-grenoble-alpes.fr), Philippe Roux (ISTerre, CNRS, Grenoble, France), Guillaume Matte (IXBlue Acoust. Systems, La Ciotat, France), and Sergey Skipetrov (LPMMC, CNRS, Grenoble, France)

Multiple scattering phenomena of acoustic waves are studied in a dense school of fish. A 3-D multibeam sonar is used to probe the backscattered intensity from a dense school fish contained in an open sea cage. The signature of wave interferences in this mesoscopic and quasi-natural media is shown by measuring coherent backscattering (CBE) effects. In particular, the static and time-resolved measurements of the CBE profile provide estimations of transport parameters such as the transport mean free path l^* and the energy velocity of diffusive waves v_e. It turns out that fish can scatter sound very strongly, allowing one to achieve a transport mean free path comparable with the wavelength of ultrasound. In this case, traditional fisheries acoustics methods are compromised by multiple scattering effects. Thus, this new approach could help to overcome issues encountered in aquaculture to evaluate the total biomass of dense schools of fish.

9:00

5aPA5. Ultrasonic wave transport in resonant emulsions made of slow or fast droplets. Benoit TALLON (I2M, Université de Bordeaux, TALENCE, France), Thomas Brunet (I2M, Université de Bordeaux, 351, cours de la libération, Bâitment A4 - I2M/APY, Talence 33405, France, thomas.brunet@u-bordeaux.fr), and John H. Page (Dept. of Phys., Univ. of MB, Winnipeg, MB, Canada)

The key transport parameters of acoustic waves travelling in disordered media are studied through ultrasonic experiments on monodisperse resonant emulsions. Through accurate measurements of both ballistic and diffusive transport over a wide range of frequencies, we show that the group velocity of the ballistic component may differ significantly from the energy velocity of the diffusive waves, in particular in the vicinity of scattering resonances. Moreover, we demonstrate the crucial role played by the refractive acoustic index of the droplets relative to the surrounding media by studying emulsions made of "slow" fluorinated oil droplets or "fast" liquid metallic droplets dispersed in a water-based gel. Our observations are successfully explained by theories based on independent scattering approximations relevant for these diluted media. Effects of droplet resonances in the diffusive regime are also discussed giving a complete study of the energy velocity of scalar waves in strongly scattering media. B. Tallon, T. Brunet, and J.H. Page, *Phys. Rev. Lett.* **119**, 164301 (2017).

9:15

5aPA6. Focusing of rarefaction wave and dynamics of cavitation zone state behind its front in two-phase cylindrical liquid layer. Valeriy Kedrinskiy (Physical Hydrodynam., Lavrentyev Inst. of Hydrodynam, Russian Acad. of Sci., Lavrentyev prospect 15, Novosibirsk 630090, Russian Federation, kedr@hydro.nsc.ru) and Ekaterina Bolshakova - Zhuravleva (Physical Hydrodynam., Lavrentyev Inst. of Hydrodynam., Russian Acad. of Sci., Novosibirsk, Russian Federation)

The evolution of the cavitation zone behind the front of a converging cylindrical rarefaction wave (RW) in a layer of two-phase distilled water is investigated numerically within the IKvanW mathematical model. The process is initiated by the generation of a shock wave (SW) exponential profile by the coaxial piston in the vicinity of the axis symmetry (R = 0 mm) in the wide range of amplitudes (U_{max} = 40–200 m/s) and time constants of exponent (10–40 μs). SW propagates along the radius of layer and reflects from a free surface (R = 60 or 100 mm) forming RW. The analysis of influence of SW parameters (at given exponent 10 μs for all amplitudes U_{max}) on the state of the cavitation zone has shown that the cavitation process is smoothly intensified for each value U_{max} in the distance ranging from 100–40 mm. Then, it drastically increases approaching the axis (interval 40–20 mm) and reaches 3500 units of the concentration (for U_{max} = 200 m/s). The total concentration becomes more than 6%. It is of interest to note that a significant effect produced by the changes in the initial parameters of state and in exponents (at U_{max} = 100 m/s): (1)—at R_0 = 1.5 μm, $k_0 = 10^{-5}$, the calculation for exponent 40 μs practically coincides with the results for 30 μs. Thus, the latter can be considered as a limit value for the accepted parameters of two-phase medium state (total concentration reaches 2%) and (2)—at R_0 = 7 μm, $k_0 = 10^{-3}$, the concentration distributions increase separately for each exponent and total concentration reaches value close to 16%.

9:30–9:45 Break

9:45

5aPA7. Influence of thermodynamics inside bubble and flow nonuniformity on weakly nonlinear waves in bubbly liquids. Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8573, Japan, kanagawa@kz.tsukuba.ac.jp), Takafumi Kamei, Taiki Maeda, and Takahiro Ayukai (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Japan)

An effect of the liquid viscosity and the thermal conductivity and nonuniform distribution of initial flow velocity on weakly nonlinear propagation of pressure waves in an initially quiescent liquid containing many small spherical gas bubbles is theoretically elucidated. Based on the derivation of the KdV–Burgers equation for a long wave and the nonlinear Schrödinger equation for a short carrier wave, the following results are obtained: (i) the installation of the energy equation affects nonlinear, dispersion, and dissipation effects; (ii) the liquid viscosity and the thermal conductivity lead to change considerably the explicit form of the coefficient in the dissipation term; and (iii) the weak nonuniform effect appears in a far field.

10:00

5aPA8. Numerical study on formation of an acoustic soliton in bubbly liquids based on weakly nonlinear wave equation. Takahiro Ayukai (Eng. Mech. and Energy, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Japan, ayukai.takahiro.sd@alumni.tsukuba.ac.jp) and Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Ibaraki, Japan)

Oscillation of microbubbles in bubbly liquids induces a dispersion effect of waves into weakly nonlinear pressure waves, and its propagation process is described by a KdV-Burgers (KdVB) equation for a long wave case. We numerically predict an evolution of waveform by solving the KdVB equation derived by Kanagawa *et al*. (2010, 2011) via a finite difference method. As a result, (i) an initial Gaussian waveform is distorted due to a nonlinear effect, and a steepening of waveform is observed on about 20 000 period; (ii) the dispersion effect is observed at a steep part of waveform, and its part becomes sharp as the initial void fraction increases; (iii) the initial waveform change into an attenuated soliton on about 100 000 period due to a dissipation effect, and the attenuation of soliton is noteworthy as the initial void fraction decreases; (iv) balance of the nonlinear, dispersion, and dissipation effects thereby forms an attenuated soliton, and its amplitude strongly depends on the value of the initial void fraction. Furthermore, the period of soliton formation depends on the type of initial waveform. Although the nonlinear, dissipation, and dispersion effects were qualitatively balanced in the KdVB equation, we conclude that these effects independently appear in the sound field.

10:15

5aPA9. Effect of a mass flux at bubble-liquid interface on propagation properties of weakly nonlinear waves. Aya Fujimoto (Eng. Mech. and Energy, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Japan, s1920933@s.tsukuba.ac.jp), Takafumi Kamei, Takahiro Ayukai (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Japan), and Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Ibaraki, Japan)

Toward the establishment and realization of medical applications accompanied with a phase change such as a drug delivery system, a formulation of the effect of mass transfer (i.e., mass flux) at the bubble-liquid interface on a ultrasound propagation process has been desired. Since a large amplitude ultrasound is utilized in such an application, the consideration of both the mass flux and the wave nonlinearity is essential. Although Fuster and Montel [*J. Fluid Mech.*(2015)] investigated linear wave propagation, nonlinear analysis has not been performed. In this paper, we theoretically clarify the effect of mass flux at the bubble-liquid interface on weakly nonlinear (i.e., finite but small amplitude) propagation of pressure waves in bubbly liquids. The effect of total evaporation or condensation as a mass flux across the bubble interface is taken into account as a simple classical model. Although various types of dissipation are installed, the effect of viscosity of gas inside the bubble is neglected. The set of basic equations for bubbly flows is composed of the conservation laws of mass, momentum, and energy, equation of motion for radial oscillations of bubble wall, equations of state, and so on. From the method of multiple scales, some nonlinear wave equations (e.g., KdV-Burgers equation) are derived. We then discuss the effects of mass flux on the propagation processes of nonlinear waves.

10:30

5aPA10. GHz ultrasound propagation with a high speed in compressible liquids containing many microbubbles. Ryosuke Akutsu (Eng. Mech. and Energy, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Japan, s1820869@s.tsukuba.ac.jp), Takanori Yoshimoto, Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Ibaraki, Japan), and Yusuke Uchiyama (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Japan)

This paper theoretically deals with weakly nonlinear (i.e., neither linear nor strong nonlinear) propagation of plane progressive quasi-monochromatic waves in an initially quiescent compressible liquid containing a tremendously large number of spherical gas bubbles on the basis of the derivation of two types of amplitude evolution equations (nonlinear wave equations). The important points are as follows: (i) the compressibility of the liquid phase, which has long been neglected, is taken into account; (ii) The incident wave frequency is very larger than an eigenfrequency of single bubble oscillations; (iii) bubbles are neither created nor annihilated; and (iv) the thermal effect is not considered and wave dissipation are then owing to liquid compressibility and liquid viscosity. By the use of the method of multiple scales with an appropriate choice of scaling relations of three-nondimensional parameters, we can derive two types of the complex Ginzburg-Landau equations (or nonlinear Schrödinger equations), where the phase velocity is larger than the speed of sound in a pure liquid.

10:45

5aPA11. Derivation of an amplitude equation for pressure wave propagation in polydisperse bubbly liquids. Reona Ishitsuka (Eng. Mech. and Energy, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Japan, s1711088@s.tsukuba.ac.jp) and Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Ibaraki, Japan)

A weakly nonlinear propagation of plane progressive pressure waves in an initially quiescent water uniformly containing small gas bubbles is theoretically investigated. In the present study, the bubbles do not coalesce, break up, appear, and disappear. The bubbles are spherical, and these oscillations are spherically symmetric. In addition, the viscosity of gas inside the bubbles and the thermal conductivities of the both phases are neglected. Although abovementioned assumptions were used in our previous studies [e.g., Kanagawa *et al*., *J. Fluid Sci. Technol.*(2010); Kanagawa, *J. Acoust. Soc. Am.* (2015)] and classical studies [e.g., van Wijngaarden, *J. Fluid Mech.* (1968)], we here incorporate polydispersity of bubbly liquids. From the method of multiple scales, an amplitude (or a nonlinear wave) equation describing a long-range propagation of waves in bubbly liquids is derived from the basic equations in a two-fluid model. By comparing the present result with the previous results assuming monodispersity, we qualitatively and quantitatively discuss an effect of polydispersity on the wave propagation process in bubbly liquids.

Invited Paper

11:00

5aPA12. Weakly nonlinear propagation of pressure waves in liquids containing translational bubbles acting a drag force. Takahiro Yatabe (Eng. Mech. and Energy, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Japan, s1920951@s.tsukuba.ac.jp), Takahiro Ayukai (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Japan), and Tetsuya Kanagawa (Eng. Mech. and Energy, Univ. of Tsukuba, Tsukuba, Ibaraki, Japan)

Weakly nonlinear (i.e., finite but small amplitude) propagation of plane pressure waves in liquids uniformly containing many spherical microbubbles is theoretically investigated. Although the effects of a translation of bubbles and a drag force acting on bubbles are incorporated, the creation, extinction, coalescence, break up, and polydispersity of bubbles are not considered. From the second order of approximation based on the method of multiple scales, the KdV–Burgers equation for a low frequency long wave is derived from the basic equations for bubbly flows composed of the conservation equations in a two-fluid model installing a virtual mass force and the bubble-dynamics equation for spherical symmetric oscillating translational bubbles. As a result, the translation of bubbles increased an effect of wave nonlinearity in a far field.

FRIDAY MORNING, 6 DECEMBER 2019 SPRECKLES, 8:00 A.M. TO 11:15 A.M.

Session 5aSA

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

Benjamin M. Goldsberry, Chair

Applied Research Laboratories, University of Texas at Austin, 10000 Burnet Road, Austin, Texas 78758

Contributed Papers

8:00

5aSA1. Immersive wave propagation experiments in a two-dimensional acoustic waveguide. Theodor S. Becker (Earth Sci., Inst. of Geophys., ETH Zurich, Sonneggstrasse 5, NO H 41.1, Zürich 8092, Switzerland, theodor.becker@erdw.ethz.ch), Nele Börsing, Dirk-Jan van Manen, Thomas Haag (Earth Sci., ETH Zurich, Zurich, Switzerland), Carly M. Donahue (Earth and Environ. Sci., Los Alamos National Lab., Los Alamos, NM), Andrew Curtis (Mathematical GeoSci., Univ. of Edinburgh, Edinburgh, United Kingdom), and Johan O. Robertsson (Earth Sci., ETH Zurich, Zürich, Switzerland)

The physical implementation of immersive boundary conditions (IBCs) allows acoustic or elastic waves to propagate seamlessly between a physical domain, such as a wave propagation laboratory, and a numerical simulation virtually enclosing the physical domain. IBCs correctly account for all wavefield interactions between both domains, including higher-order long-range scattering. In this contribution, IBCs are physically implemented in a two-dimensional (2-D) acoustic waveguide. The boundary surrounding the waveguide is densely populated with hundreds of loudspeakers that apply the necessary boundary conditions. The required signals to be injected at the boundary are predicted in real-time by (1) measuring the pressure field and its gradient on two acoustically transparent auxiliary surfaces of microphones inside the waveguide and (2) extrapolating the wavefields to the boundary by evaluating a Kirchhoff-Helmholtz integral using a low-latency, FPGA-enabled data acquisition, computation and control system. Here, we demonstrate the first real-time, 2-D physical immersive wave propagation experiments. We present the setup, as well as a suite of experiments that demonstrate the ability of IBCs to actively suppress broadband incident fields at the boundary of the waveguide and to correctly reproduce all orders of wavefield scattering between the physical experiment and the numerical simulation.

8:15

5aSA2. Overview of the fixed base correction method for modal testing on dynamic boundary conditions. Peter Kerrian (ATA Eng., Inc., 13290 Evening Creek Dr. South, San Diego, CA 92128, peter.kerrian@ata-e.com) and Kevin Napolitano (ATA Eng., Inc, San Diego, CA)

Modal tests are important steps in the validation of structural dynamic finite element models of test articles. Often, inadequate boundary conditions require more effort to be spent correlating the boundary condition or support structure rather than the test article. Recently, boundary condition correction methods, such as the fixed base correction method, have been developed to extract fixed based mode shapes for test articles on dynamic test fixtures. This paper presents an overview of the fixed base correction method and provides examples of the implementation to academic and representative test articles.

8:30

5aSA3. Application of the fatigue damage spectrum to accelerated vibration testing. Odissei Touzanov (Acoust., The Penn State Univ., University Park, PA) and Manton J. Guers (Structural Acoust., Penn State Appl. Res. Lab., PO Box 30, State College, PA 16804, mjg244@psu.edu)

Vibration testing is an important part of product validation in many industries. However, the time to conduct a vibration test is an important

consideration, especially in industries where designs need to be validated quickly. In this work, the Fatigue Damage Spectrum (FDS) methodology was validated against results presented in the literature and then applied to a vibration test profile based on road load data. An experimental validation of the code was performed using an electrodynamic shaker. It was demonstrated that the FDS methodology can cut the testing time in half and produce reasonable results based on the criteria of maximum crack length of the specimens.

8:45

5aSA4. Design simulations for a combined shaker-acoustic vibration test technique. Ryan Schultz (Structural Dynam., Sandia National Labs., 1701 Singletary Dr. NE, Albuquerque, NM 87112, rschult@sandia.gov)

A novel approach of combining multiple shaker and acoustic inputs can achieve a high-fidelity ground vibration test. Additionally, supplementing acoustic inputs with shakers enables higher response levels which is possible with shakers alone. As this is a new testing concept, exploring test design effects using simulations is useful. Here, models and simulation are utilized to explore how combining shaker and acoustic inputs can improve response levels and reduce input requirements while maintaining test accuracy. The derivation of acoustic and shaker inputs is explored using different control or input estimation techniques. In particular, the correlation requirements between shaker and acoustic inputs is examined using a Kronecker product formulation of the control problem, which enables specific inputs to be uncorrelated. This simulation-based test design approach provides useful guidance in the design of combined shaker-acoustic vibration tests. [Sandia National Laboratories is a multimission laboratory managed and operated by National Technology & Engineering Solutions of Sandia, LLC, a wholly owned subsidiary of Honeywell International, Inc., for the U.S. Department of Energy's National Nuclear Security Administration under Contract No. DE-NA0003525.]

9:00

5aSA5. A comparative analysis of fitness flooring buildups designed for heavy weight impacts. Jessica Scarlett (Getzner USA, 8720 Red Oak Blvd, Ste. 400, Charlotte, NC 28217, jjstamey@gmail.com)

A comparitve analysis of fitness flooring builups designed for heavy weight impacts. Multiple combinations of polyurethane foam and rubber topping layers were laboratory tested for sound and vibration performance. Variables included polyurethane density and thickness and rubber topping thickness, mass of weights dropped, and height of weights dropped. The review of data aims to identify ideal buildups for specific weight impacts and drop heights.

9:15

5aSA6. High bandwidth airborne acoustic detection system (HBADS) for circular synthetic aperture acoustic imaging of canonical ground targets. Steven S. Bishop (US Army RDECOM CERDEC Night Vision and Electron. Sensors Directorate, Ft. Belvoir, VA 22060, steven.s.bishop4.civ@mail.mil), Timothy R. Moore, Peter Gugino, Brett Smith (US Army RDECOM CERDEC Night Vision and Electron. Sensors Directorate, Ft. Belvoir, VA), Kathryn P. Kirkwood, and Murray S. Korman (Phys. Dept., U.S. Naval Acad., Annapolis, MD)

High Bandwidth Acoustic Detection System (HBADS) is an active acoustic sensor technology developed by the US Army's Night Vision and Electronic Sensors Directorate. The stripmap airborne synthetic aperture acoustic (SAA) array (mounted on an all-terrane vehicle) contains two horizontal rows of eight evenly spaced microphones oriented in a side scan configuration. An audio speaker (located near the array center) transmits a linear FM chirp having a 2 kHz to 15 kHz effective bandwidth. Experiments are performed with the vehicle in motion to image canonical ground targets in clutter or foliage. A laptop computer controls the 16 channel data acquisition system. The system includes an inertial navigation system (two differential GPS antennas), a wheel coder, a camera, and a unit for measuring temperature, pressure, and humidity. Experiments include imaging spherical targets located on the ground in a grassy field and similar targets camouflaged by natural vegetation along the roadside using both stripmap SAR and circular path C-SAR radar-like configurations. Typical parameters are chirp pulse = 10 or 40 ms, slant range resolution $c/(2*BW)$ = 1.3 cm (air sound speed c $\approx$ 340 m/s), microphone diameter D = 2 cm, (azimuthal resolution is D/2), and maximum vehicle speed $\approx$ 2 m/s.

9:30–9:45 Break

9:45

5aSA7. Noise source modifications to enhance performance of an acoustic modulation system. Melissa A. Hall (Structural Validation Branch, Air Force Res. Lab., 2790 D St., Bldg 65, Wright-Patterson AFB, OH 45433, melissa.hall.7@us.af.mil)

The ability to simulate extreme flight environments of aerospace structures, such as high-intensity thermal acoustic loading, is achieved through the use of a progressive wave tube facility. This valve modulation system, used to create the random acoustic wave, behaves like an electromagnetically driven spring-mass system that uses electro-pneumatic, vibrating-vane air valves as the noise source. Each valve requires an electrical input and reaction with a magnetic field to generate the force necessary to overcome the spring stiffness of the outer portion of the valve. This force, driven by a current carrying wire, opens and closes the modulation slots, creating airflow pressure pulses necessary to generate higher sound pressure levels. Several physical characteristics of the noise source can be modified to generate a higher force on the spring section of the valve. This work documents the increase in the overall sound pressure level output of the noise source by varying the cross-sectional area and gage size of the current carrying wire, documenting performance of materials which strengthen the magnetic field, and by comparing the frequency about which the narrow band drive signal was centered. Based on the results of these comparisons, an optimal combination of physical characteristics will be discussed.

10:00

5aSA8. A numerical study of blast wave mitigation using volumetric diffusors. Kyle G. Dunn (Cold Regions Res. and Eng. Lab., Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755, kyle.g.dunn@usace.army.mil), Michael B. Muhlestein (Cold Regions Res. and Eng. Lab., Engineer Res. and Development Ctr., Hanover, NH), and Michelle E. Swearingen (Construction Eng. Res. Lab., Engineer Res. and Development Ctr., Champaign, IL)

We explore the use of large-scale volumetric diffusors to reduce the power of a blast wave observed far away from a point source in a given direction. The blast wave propagation is approximated as being linear and modeled using a three-dimensional finite difference time domain method. The volumetric diffusor under consideration consists of many rigid scatterers and is intended to allow energy to propagate through while decreasing the maximum impulsive pressure. The optimization of rigid scatterer positions is used to achieve our goal of 5–7 dB within an arc of at least 30 deg. An overview of the model and methods will be presented along with preliminary results showing the effectiveness of our approach.

10:15

5aSA9. Flexural vibration of steel beam with locally resonant phononic crystal: An experimental investigation. Tongjun Cho (Yonsei Univ., Yonseino 50 Seodaemun-gu, Seoul 120749, South Korea, tjcho@yonsei.ac.kr), Jewoo Choi, Sang Geun Bae, and Hyo Son park (Yonsei Univ., Seoul, South Korea)

The effect of locally resonant phononic crystal on the bending wave propagation behavior of a full-scale steel beam structure was experimentally investigated. The unit cell of the phononic crystal was designed with a composition comprising a tungsten mass, a silicone rubber spring, and a plaster matrix. The vibration resonances of the coated tungsten sphere of the phononic crystal units were evaluated by an analytical model and experimental measurements. The frequency response functions of the beam structure indicated that *modal* energy of the *bending* wave vibration is significantly reduced due to the locally resonant mechanism of the attached phononic crystal units. The local resonance bandgap observed in the experiment suggest that phononic crystals can be used for a method of flexural vibration control of beam structures.

10:30

5aSA10. On improving sound beam formation using the special acoustic lens in sound extinguisher. Bong Young Kim (Commun. Eng., Soongsil Univ., 21-1, Garak-ro 23-gil, Songpa-gu, #203, Seoul 05669, South Korea, bykim8@ssu.ac.kr) and Myungjin Bae (Commun. Eng., Soongsil Univ., Seoul, South Korea)

New fire prevention technologies have become necessary due to various fire fighting environments. Sound fire extinguishers can be a new alternative technique to suppress fire by using sound characteristics. We have newly proposed a special acoustical lens to improve the sound beam forming of a sound extinguisher. In this paper, the characteristics of the sound beam formed by the special acoustic lens and the sound beam are verified through experiments. While the wind speed was measured in the sound beam, when the reverse phase sound was supplied, the wind speed was lost and the sound amplitude was reduced by about 20 dB, and the sound velocity of the sound beam was measured to move the medium at a constant cycle with a very large pressure change. Also, the sound level on the propagation path was measured to be 4 dB attenuation up to 30 cm, which is 3 times of 10 cm, and it was confirmed that sound energy was effectively transmitted to 70 cm. As a result of this experiment, the special sound lens of the sound extinguisher emits the resonant sound as the sound source of the plane sound source, so that the sound energy is concentrated and transmitted to the flame effectively.

10:45

5aSA11. A study on compatibility of rhythm sound with existing sound in Clackson. WonHee Lee (Commun. Eng., Soongsil Univ., Hyeongnam Eng. Hall #1212, Sangdo-dong, Dongjak-gu, Seoul 06978, South Korea, wbluelovew@gmail.com) and Myungjin Bae (Commun. Eng., Soongsil Univ., Seoul, South Korea)

The sound of a car 's clackson was developed and used 100 years ago, but can cause traffic noise to pedestrians and drivers, which can be harmful. We have previously proposed a new familiar Clackson sound with a different rhythm or tempo using the existing Clackson. However, since the existing Clackson sound is needed in emergency or special cases, this study newly proposed a compatible Clackson sound method. First of all, pressing the center part of the handle of the driver's seat while driving starts a rhythmic clack sound, on the other hand, after a short touch, you can press and hold down to use the existing Crackson sound. We usually use a human-friendly Klaxon sound that is designed considering 100 people's hearing fatigue, brain wave test, stress index, etc., and when using the existing Klaxon sound in the case of emergency. Therefore, the proposed Clackson sound can be used without any major change in all the cars on the market, so that it is possible to minimize the opponent's retaliation during driving.

FRIDAY MORNING, 6 DECEMBER 2019 CROWN, 8:00 A.M. TO 12:00 NOON

Session 5aSC

Speech Communication: Speech Articulation (Poster Session)

Matthew Faytak, Chair

Linguistics University of California, Los Angeles, 31-19 Rehabilitation Center, 1000 Veteran Avenue, Los Angeles, California 90095

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors in this session an opportunity to see other posters, contributors 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 to 12:00 noon.

Contributed Papers

5aSC1. Tongue root configuration during Seoul Korean stops: An ultrasound study. Suzy Ahn (Dept. of Linguist, UCLA, 3125 Campbell Hall, Los Angeles, CA 90095, suzyahn@ucla.edu) and Harim Kwon (Dept. of English, George Mason Univ., Fairfax, VA)

Tongue root advancement facilitates voicing during stop closure by enlarging the supralaryngeal cavity volume (Westbury, 1983). In a recent ultrasound study, Ahn (2018) reports that tongue is indeed more advanced during voiced than voiceless stops both in English and Brazilian Portuguese, suggesting that the articulatory adjustment aligns more with the abstract laryngeal distinctions than their acoustic implementation—the "voiced" stops are typically not phonated in English but in Brazilian Portuguese. This study, using ultrasound, compares tongue positioning during Seoul Korean (SK) stops of three laryngeal categories: lenis, fortis, and aspirated. All three categories are voiceless phrase-initially, with the lenis being voiced intervocalically (Jun, 1993). This study asks whether (1) SK lenis, fortis, and aspirated stops have different tongue configurations when none are phonated and (2) the intervocalic voicing of lenis stops leads to tongue root advancement. Nine native SK speakers recorded phrase-initial and intervocalic stops. Results: lenis, fortis, and aspirated stops did not show different tongue position in phrase-initial positions. In intervocalic positions, lenis stops, acoustically voiced during closure, did not show more tongue root advancement than other types of stops. These results suggest that SK speakers use tongue positioning neither for laryngeal contrast nor as an adjustment for allophonic voicing of intervocalic lenis stops.

5aSC2. Two applications of movable mandible for physical models of the human vocal tract. Takayuki Arai (Information and Commun. Sci., Sophia Univ., 7-1 Kioi-cho, Chiyoda-ku, Tokyo 102-8554, Japan, arai@sophia.ac.jp)

The first application of a movable mandible for physical models of the human vocal tract is based on the 2017 static-model, the vocal-tract configuration of which was set to produce the vowel /a/. Two parts of the model, upper and lower jaws, are combined with a miniature robot hand unit to

achieve articulatory movements for visual cues in a speech perception study. The second application is based on the 2018 dynamic-model, which is physically able to produce human-like speech sounds with a flexible material. Because portions of the articulators including the top surface of the tongue are made of a gel-type material, a user can manipulate the shape of the tongue and articulate different vowels and a certain set of consonants for educational purposes, for example. However, the mandible of the model is fixed, making it difficult to manipulate different sounds with different jaw openings, such as high versus low vowels. Therefore, a new model was developed by adding an additional mandible mechanism to the 2018 model. The mandible was designed to move between the open and closed positions by creating an arc-shape rail. As a result, various speech sounds with a flexible-tongue and movable mandible can be easily produced.

5aSC3. Post-collection ultrasound-audio synchronization. Sarah Bakst (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, 1500 Highland Ave., Madison, WI 53705, sbakst@wisc.edu) and Susan Lin (Linguist, Univ. of California Berkeley, Berkeley, CA)

Speech research using lingual ultrasound often requires both the ultrasound images as well as audio from the corresponding audio speech signals, but synchronization of these signals is not always available. We propose that periods of matching rates of change in the two signals could be used to align articulatory and acoustic signals where synchronization is impossible or would otherwise benefit from verification. In this study, we analyze pre-synchronized ultrasound and audio recordings of English speakers reading words. For each recording, we calculated the articulatory change as the change in pixel brightness values over time and the acoustic change as the change in Mel Frequency Cepstral Coefficient representations of the audio, and then calculated the degree of correlation between the acoustic and articulatory change over a window shifting through the recording. Then, we deliberately offset the signals in increments of 5 ms to verify that the known synchronization results in the best correlations. We manipulated several other variables, and preliminary results suggest that shorter window lengths and analyzing correlations only during detected speech result in the most accurate alignments. Analysis is ongoing to determine whether duration of correlation (number of windows with high r-values) or overall degree of correlation (median r-values) leads to the most accurate alignments.

5aSC4. Beatrhyming probes the nature of the interface between phonology and beatboxing. Reed Blaylock (Univ. of Southern California, 1150 W 29th St. Apt 4, Los Angeles, CA 90007, reed.blaylock@gmail.com) and Ramida Phoolsombat (Univ. of Southern California, Los Angeles, CA)

Beatboxing—a form of vocal music made with percussive sounds—has been shown to have its own set of sounds and grammatical properties. The development of beatrhyming—an art form in which beatboxing and speech are performed simultaneously by a single individual—presents the first opportunity to explore what happens when speech competes with a similar vocal task. In beatrhyming, beatboxing sounds often replace word onset consonants; for example, the word "float" /flot/ in beatrhyming can be pronounced as [pf'lot], with a PF Snare replacing the initial /f/ . In principle, the speech sound could be replaced by any beatboxing sound; however, English beatrhyming data from YouTube show that in many cases, the beatboxing sound shares articulatory properties (such as place of articulation) with the speech sound which it replaces. Beatrhyming is one of the only behaviors that forces an interaction between speech and another complex grammatical system of sounds; this makes beatrhyming a unique venue for exploring the relationship between speech phonology and other related aspects of human cognition.

5aSC5. Evaluation of excised porcine larynx frequency changes due to changing vocal fold elastic properties. Garret Burks (Mech. Eng., Virginia Tech, 230 Goodwin Hall, Blacksburg, VA 24061, garretb7@vt.edu), Alexander Leonessa (Mech. Eng., Virginia Tech, Blacksburg, VA), and Raffaella De Vita (Biomedical Eng. and Mech., Virginia Tech, Blacksburg, VA)

The work presented in this study investigates the impact of changing vocal fold elastic properties on the frequencies of vibration during phonation. Excised porcine larynx tests were conducted to examine the relationship between porcine vocal fold anterior-posterior strain and the resulting changes in the phonation frequency. During each test, two high speed cameras (4000 FPS) and a microphone (44100 Hz) were used to record changes in vocal fold dynamics while a servo motor stretched the tissue in the anterior-posterior direction. A digital image correlation tracking method was then used to measure anterior-posterior strain on the exposed porcine vocal fold superior surface. Strain measurements were related to changes in vocal fold elasticity based on previously reported mean elastic parameters which describe porcine vocal fold elasticity from 0%–40%. Additionally, an aeroelastic model of phonation which incorporates the measured vocal fold geometry, elastic properties, and air flow characteristics was implemented to further study the impact of changing porcine vocal fold elasticity on the vibration frequency. Phonation was first observed at a strain of 0.18 mm/mm with a fundamental frequency of 246 Hz and increased to 404 Hz at a strain of 0.35 mm/mm. Airflow conditions were kept constant during all experimental tests.

5aSC6. An electromagnetic articulography-facilitated deep neural network model of tongue contour detection from ultrasound images. Wei-Rong Chen (Haskins Labs., 300 George St. STE900, New Haven, CT 06511, chenw@haskins.yale.edu), Mark Tiede (Haskins Labs., New Haven, CT), Jaekoo Kang (Speech Lang. Hearing Sci., City Univ. of New York, Long Island City, NY), Boram Kim (Linguist, City Univ. of New York, New York, NY), and D. H. Whalen (Haskins Labs., New York, NY)

Ultrasound imaging is a non-invasive technique for the measurement of the tongue in speech. Recent advancements in analytical edge detection algorithms and deep learning methods have improved tongue contour segmentation. However, most edge detection algorithms require user input as initialization "seeds" and accuracy can drift as segmentation subsequently proceeds. Deep learning in ultrasound tongue contour tracking, on the other hand, requires large, manually labelled training data, has poor spatial resolution and does not generalize well to images acquired by ultrasound machines outside the training set. Here, we demonstrate an approach combining both edge detection and deep learning in automatic tongue contour tracking in ultrasound images of the tongue, aided by synchronously acquired electromagnetic articulometry (EMA) data. A deep learning model was trained by edge detection seeded by tongue locations from EMA with minimal human intervention. Spatial and temporal constraints for tongue contours were learned simultaneously using a three-dimensional convolutional neural network. Finally, the deep neural network inferred tongue contour in low resolution was passed through additional edge detection to refine the contour in higher resolution. Our preliminary results demonstrate the proposed architecture improved accuracy compared to using analytical edge detection or deep learning alone.

5aSC7. Overlap and sonority in complex onsets in Georgian. Caroline Crouch (Dept. of Linguist, Univ. of California, Santa Barbara, South Hall 3432, Santa Barbara, CA 93106, crouch@ucsb.edu) and Argyro Katsika (Linguist, Univ. of California, Santa Barbara, Santa Barbara, CA)

This study investigates the relationship between the sonority shape of complex syllable onsets and articulatory overlap between the constituent consonants in Georgian (Kartvelian, Georgia; geo). Sonority sequencing and related phonological phenomena can be analyzed as epiphenomena related to perceptual cue preservation. The current study builds on previous research on Georgian that shows that order of place of articulation affects degree of overlap between stop-to-stop sequences and relates this back to issues of recoverability of segments within the sequence, and expands on this research to include complex syllable onsets with segmental makeups other than, but including, stop-to-stop sequences. Specifically, we use electromagnetic articulography (EMA) to examine the movements of the tongue, lips, and jaw during the production of Georgian words that cross three sonority shapes (rise, fall, and plateau) with front-to-back and back-to-front orders of place of articulation. Preliminary analyses confirm that there is less overlap in front-to-back clusters than in their back-to-front counterparts. It is also shown that sonority shape significantly affects overlap measures in interaction with order of place of articulations. This points towards a systematic relationship between sonority and overlap, which is discussed

within the context of perceptual cue preservation and recoverability [Work supported by NSF.]

5aSC8. Co-speech movement behavior in conversational turn-taking. Samantha G. Danner (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, sfgordon@usc.edu), Jelena Krivokapic (Linguist, Univ. of Michigan, Ann Arbor, MI), and Dani Byrd (Linguist, Univ. of Southern California, Los Angeles, CA)

Co-speech movements like those of the hands, head and face are thought to be important to the seamless turn-taking that is a hallmark of conversational interaction. We hypothesize that the spatiotemporal characteristics of co-speech movements vary as a function of turn-taking context (e.g., interruptions, felicitous floor exchanges) and/or a participant's conversational role (speaker/listener). Though there is very little prior research in this area, work on manual gestures has shown that the rate of gesture (*gestural density*) varies contextually. In this study, we examine *gestural density* of head and brow gestures of interacting speakers considering two factors: (1) type of conversational turns (compared to non-turn-adjacent speech) and (2) participant conversational role. We analyze four dyads participating in a naturalistic conversational speech task recorded with magnetometer movement-tracking of lips, maxilla, brows, jaw, and head. Linear mixed effect models tested whether gestural density of head-nods and brow-raises differs among turn type and/or conversational role. Significant effects and interactions for both brow and head movement indicate that co-speech gestural density is jointly modulated by turn type and role. We find an examination of co-speech movement to be a fertile field of exploration in the study of interactional prosody and discourse. [Work supported by NIH.]

5aSC9. Co-speech laughter in a conversational speech task. Nynaeve Perkins Booker (Linguist, Univ. of Southern California, Los Angeles, CA), Cheyenne M. LaRoque (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, laroque@usc.edu), Samantha G. Danner (Linguist, Univ. of Southern California, Los Angeles, CA), Jelena Krivokapic (Linguist, Univ. of Michigan, Ann Arbor, MI), and Dani Byrd (Linguist, Univ. of Southern California, Los Angeles, CA)

Laughter plays a role in the natural flow of conversation. This study examines four dyads participating in a semi-structured collaborative conversational speech task to assess the effects of participant role—listener versus speaker—and the location of laughter within a speech turn on laughter distribution and duration. We hypothesize that speakers' laughter will be more constrained than listeners' due to its limitation to speech turn boundaries and its accommodation of a speaker's ongoing speech planning requirements. Additionally, based on prior findings on acoustic phrase-final lengthening in speech prosody, we expect that laughter at the end of a speech turn will be longer than turn-initial or -medial laughter. Our results demonstrate substantial individual variability, with a majority of participants having longer laughter duration as listeners than as speakers, in part, we propose, due to laughter's use as a form of backchanneling. Participants also demonstrate longer durations of turn-initial laughter than turn-final or turn-medial laughter. This unexpected result may be due to overlapping speech and laughter that is often seen at the end of speech turns. Our results suggest that the patterning of laughter with respect to speech turns may help structure interactive speech discourse. [Work supported by NIH.]

5aSC10. Human beatboxing : A multi-instrumental pilot. Alexis Dehais Underdown (Laboratoire de Phonétique et Phonologie (UMR 7018 - CNRS/Sorbonne-Nouvelle), 61, rue de la folie régnault, Paris 75011, France, alexis.underdown@gmail.com), Paul Vignes, Lise Crevier-Buchman, and Didier Demolin (Laboratoire de Phonétique et Phonologie (UMR 7018 - CNRS/Sorbonne-Nouvelle), Paris, France)

This comparative study aims to describe laryngeal, acoustic, and aerodynamic characteristics of instrumental imitations that is the beatboxed classic kick drum [pã] and snare drum [pëÀf] (PF-snare) by one artist. Beatboxed sounds were produced in isolation and in beatboxed patterns. Aerodynamic (intraoral pressure, oral airflow, and nasal airflow), acoustic, electroglottographic, and laryngoscopic data were acquired. Laryngeal and aerodynamic data show that the classic kick drum is produced as a bilabial voiceless glottalic egressive plosive [p] or as a bilabial voiceless glottalic egressive affricate [pè¡]. The PF-snare was produced as a labial voiceless glottalic egressive affricate [pëÀf]. Sounds in isolation were shorter than those in a pattern. Acoustic analysis revealed high amplitude burst for all realizations that is consistent with ejective realization (i.e., high increasing intraoral pressure). Spectral analysis shows a peak in the low frequencies for [p] as well as for [pëÀf]'s onset but when turning to the fricative offset [f], we noted a peak in high frequencies and a flattening spectrum. Based on the acoustic and aerodynamic data, we discuss the coordination of the articulators and the planification of articulatory commands of the classic kick drum and the PF-snare drum.

5aSC11. Respiration in speech: Control, global and local effects. Didier Demolin (Linguist, LPP Paris 3, 19 rue des bernardins, Paris, France, didier.demolin@sorbonne-nouvelle.fr), Shi Yu (Linguist, LPP Paris 3, Paris, France), and Sergio Hassid (HûĈpital Erasme, Université Libre de Bruxelles, Bruxelles, Belgium)

Breathing is essential to man's ability to speak. The respiratory bellows provide the power to the vocal apparatus. Expiration in speech often continues until lung volume decreases below functional residual capacity. Speakers appear to achieve a compromise between ventilatory and speech demands on flow rates. The question addressed in this presentation is how is the relative constancy of subglottal pressure (Ps) achieved despite the continual change of relaxation forces? During speech an extra 6/10 hPa is sustained above atmospheric pressure to provide the energy to speak. As showed in the literature, external and internal intercostal muscles are the most important to regulate Ps, not the diaphragm. In speech, Ps is sustained by the expiratory muscles after the recoil of the lungs tissues. Global effects show, from data obtained with direct measurements, that the slight Ps declination in sentences is largely due to the system's compliance and changes in glottal and oral resistance. Local effects such as those on trills and stress, where increases of 1–2 hPa are observed, are also accounted for by modifications in glottal and oral resistance. Our results show that Ps and F0 that are regulated by different controls (spinal and cranial nerves) interact in complex ways.

5aSC12. Flap articulation and lowered fourth formant. Matthew Faytak (Dept. of Linguist, Univ. of California, Los Angeles, UCLA 3125 Campbell Hall Box 951543, Los Angeles, CA 90095, faytak@ucla.edu), Jacob Aziz, Phillip Barnett, Jinyoung Jo, Jennifer Kuo, G. Teixeira, Joy Wu, Z. L. Zhou, and Patricia Keating (Linguist, Univ. of California, Los Angeles, Los Angeles, CA)

Speakers of North American English are known to use a variety of tap/flap articulations depending on phonetic context (Derrick and Gick, 2011); it is also known that NAE taps/flaps are sometimes associated with a greatly lowered F4 frequency (Warner and Tucker, 2017). It has been less clear whether only certain articulatory variants show this acoustic effect. Since retroflex stops are also associated with lowered F4 (Blumstein and Stevens, 1975), we predict that flap retroflexion is associated with lowered F4. To test this prediction, synchronized ultrasound and audio recordings were made of words containing /t, d/ in a variety of contexts known to give rise to tap/flap variants. Based on visual inspection of ultrasound videos, these were coded as one of four articulatory variants (low tap, high tap, up flap, down flap: Derrick and Gick, 2011); formant frequencies were extracted from the audio at several timepoints relative to the tap/flap. Preliminary results from one speaker support the hypothesis: high taps and down flaps (variants with initial retroflexion) show an F4 drop into the consonant, while high taps and up flaps (variants with final retroflexion) show an F4 rise out of the consonant.

5aSC13. Voice actors simulating small vocal tracts: A study using 3-D ultrasound. Colette Feehan (Linguist, Indiana Univ., Bloomington, 107 s Indiana Ave., Bloomington, IN 47404, cmfeehan@indiana.edu)

Voice actors are an interesting population for linguistic study because their professional field requires them to perform complex vocal tract manipulations in order to consistently and believably convey many different character types. Previous investigations have looked at how voice actors manipulate laryngeal setting and voice quality, to portray specific types of characters in animation (Teshigawara and Murano, 2004; Starr, 2015), but investigations of articulatory manipulations employed by voice actors are rare. This study uses 3-D ultrasound and acoustic analyses to compare the different kinds of articulatory and phonatory strategies that different actors use in order to approximate a smaller vocal tract and achieve a similar ã childlike' percept. Preliminary analysis shows that one amateur actor relies on manipulation of articulatory setting by implementing hyoid bone raising, gesture fronting, and tongue grooving, whereas one professional actor relies more on manipulation of laryngeal structures and prosody. Despite these differences in approach, the two actors still achieve similar child-like percepts. This poster will compare strategies from two female and two male subjects as well as describe within-subject variation across each actor's adult and imitated child voices.

5aSC14. McGurk doesn't work: Individual differences and task demands explain the McGurk illusion. Laura M. Getz (Psychol. Sci., Univ. of San Diego, 5998 AlcalûÀ Park, Serra Hall 158, San Diego, CA 92110, lgetz@sandiego.edu) and Joseph C. Toscano (Psych., Villanova Univ., Villanova, PA)

Visual speech cues play an important role in speech recognition, and the McGurk effect is a classic demonstration of this. In the original McGurk and MacDonald (1976) experiment, 98% of participants reported an illusory "fusion" percept of /d/ when listening to the spoken syllable /b/ and watching the visual speech movements for /g/. However, recent work (e.g., Mallick *et al.*, 2015) shows that subject and task differences influence the proportion of fusion responses. In the current study, we varied task (forced-choice versus open-ended), stimuli (synthetic versus naturally produced), design (mixed versus blocked audio-visual and single-modality trials), and data collection environment (lab versus Mechanical Turk) to investigate the robustness of the McGurk effect. Across all conditions, we found a low number of fusion responses. Rather than a robust perceptual illusion, we therefore argue that the McGurk effect is a product of individual differences and task demands.

5aSC15. Accurate detection of spatially calibrating laser points during transnasal, fiberoptic, laryngeal high-speed videoendoscopy using support vector machine. Hamzeh Ghasemzadeh (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd. Oyer Speech & Hearing, Rm. 211-C, East Lansing, MI 48824-1220, ghasemza@msu.edu), Dimitar Deliyski (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), Robert E. Hillman, and Daryush D. Mehta (Dept. of Surgery, Massachusetts General Hospital–Harvard Med. School, Boston, MA)

Performing calibrated spatial measurements of vocal fold structures (including pathology) and kinematics during phonation is essential for developing patient-specific voice production models and better quantifying treatment outcomes. Unfortunately, conventional videoendoscopic images cannot provide such measurements. However, adding laser-based fiducial markers to the field of view (FOV) can provide measurements in the coronal and transverse planes. Accurate detection of the laser points during *in vivo* recordings is a pre-requisite of such a measurement. This study presents a novel method based on machine learning for detection of the laser points. A custom-developed transnasal, fiberoptic endoscope projected a pattern of 7×7 green laser points on the FOV during sustained vowel tokens, which was recorded with a color high-speed video camera at 6000 frames per second. Analysis of the images showed that each laser point travels along a unique and deterministic trajectory. Based on this, the classification score of a support vector machine was employed for maximizing the posterior probability and finding the most probable laser point in each trajectory. Classification features were tailored to account for geometrical shape and gradient profile of laser points. Additionally, features were extracted from red, green, and blue color channels for accurate distinction between laser points and reflection light.

5aSC16. A real-time magnetic resonance imaging study of cross-speaker variability in the production of /ɹ/. Bianca P. Godinez (Linguist, CSULB, 1250 Bellflower Blvd, Long Beach, CA 90840, bianca.godinez@student.csulb.edu), Asterios Toutios, and Shrikanth S. Narayanan (Elec. and Comput. Eng., Univ. of Southern California, Los Angeles, CA)

There is a longstanding interest in understanding the production mechanisms of the American English (AE) rhotic, /ɹ/ especially given both the wide intraspeaker variability and interspeaker heterogeneity in its articulation, and the resultant acoustics. A diversity of tongue shapes, from those creating a more anterior tongue-tip ("retroflex") constriction to those with a more posterior bunched tongue dorsum constriction, has been observed with notable intraspeaker variation as a function of context; more specifically, a tendency for retroflexion preceding low and/or back vowels. The present study aims to analyze AE /ɹ/ production using real-time MRI videos (83 frames/s) where the target sound was produced in several contexts by 32 American English speakers (16 F/16 M). Initial findings in symmetric VCV production reveal that some of these speakers exclusively use a "retroflex" strategy, some a "bunched" strategy, and others both strategies, consistent with the previously observed tendencies. This observed variability, and the fact that multiple types of static and dynamic imaging data are available from the same speakers, makes this dataset ideal for further probing /ɹ/ production strategies in conjunction with speaker-specific characteristics of vocal tract structure, such as palate shape, and function, such as inter-articulator coordination. [Work supported by NIH and NSF.]

5aSC17. Variability is variable: Individual differences in articulatory variation. Sarah Harper (Linguist, Univ. of Southern California, 3601 Watt Way, Grace Ford Salvatori 301, Los Angeles, CA 90089, skharper@usc.edu) and Louis Goldstein (Linguist, Univ. of Southern California, Los Angeles, CA)

The extent of the variability observed for a specific articulatory parameter or dimension differs across segments in a language (Amerman *et al.*, 1970 and Vaissiere, 1983). Previous research suggests that these differences are related to the contribution of the parameter to the maintenance of phonological contrast (e.g., Keating, 1990) or to the effect of the parameter on the hypothesized goal of a phonological unit (Nieto-Castanon *et al.*, 2005). In this study, we extend this research to individual differences in articulation by examining whether individual speakers differ in the extent of the variability they exhibit along a given articulatory parameter. Tongue tip kinematics from 40 American English speakers in the Wisconsin X-Ray Microbeam Corpus (Westbury, 1994) were analyzed to determine the variation in the location and degree of the tongue tip constriction gesture for each speaker in a subset of the consonants (/s/, /l/, /ɹ/, and /t/). Results indicate that the extent of the variability observed along these constriction dimensions varies markedly both between and within speakers, with speakers differing not only in the amount of variation they exhibit along each dimension but also in which dimension exhibits more variability. [Work supported by NIH.]

5aSC18. Lingual articulatory configurations of Japanese devoiced vowels. Rion Iwasaki (Speech-Language-Hearing Sci., City Univ. of New York, 365 Fifth Ave., New York, NY 10016, riwasaki@gradcenter.cuny.edu), D. H. Whalen, Kevin Roon (Speech-Language-Hearing Sci., City Univ. of New York, New York, NY), and Jason A. Shaw (Dept. of Linguist, Yale Univ., New Haven, CT)

High vowels in Tokyo Japanese are typically devoiced between voiceless obstruents, but there is controversy about whether they can also be deleted entirely or not. Deciding this issue is important because vowel deletion would have implications for Japanese phonology and phonotactics such as consonant clusters in word-initial position and their syllabification. Here, we use ultrasound to examine whether devoiced vowels have the same tongue configurations as those of their voiced counterparts (i.e., they are simply unphonated) or not (i.e., they are deleted). For comparison, this study

also examines whispered speech. The whispered vowels are assumed to be unphonated and presumably have the same tongue configurations as the voiced vowels. Native speakers of Tokyo Japanese produced word pairs contrasting in the voicing realization of a high vowel. The comparison between devoiced, voiced, and whispered vowels was made by quantifying differences in the tongue contour traced from respective ultrasound frames. Several consonantal conditions of devoicing were considered. Preliminary results indicate that devoiced vowels might have different tongue configurations from those of voiced and whispered vowels in certain consonantal conditions.

5aSC19. Articulatory characteristics of focus in Korean. Jiyoung Jang (Linguist, Univ. of California Santa Barbara, South Hall 3432, Santa Barbara, CA 93106, jiyoung@ucsb.edu) and Argyro Katsika (Linguist, Univ. of California Santa Barbara, Santa Barbara, CA)

The phonetic correlates of prominence in head prominence languages (e.g., English), where prominence is signaled by stress and pitch accent, are extensively studied. However, the phonetic profile of prominence in edge prominence languages, such as Korean, that realize prominence by phrasing is not well understood. Relevant work has mainly focused on the acoustic domain and boundary-related phenomena. The present study assesses the articulatory correlates of prominence in Seoul Korean by the means of electromagnetic articulography (EMA). Specifically, we examine the effect of presence of focus (contrastive focus versus absence of focus) and position of focus (initial versus final) on the consonant constriction gestures of utterances consisting of two four-syllable long accentual phrases (APs). Preliminary results suggest that constriction gestures are longer, faster, and, although weakly supported, larger in contrastive focus than in the unfocused condition. These effects are found on the onset consonants of the first two syllables of the focused AP when the latter is utterance-initial, but extend to the coda of the second syllable when focus is on the final AP. A theoretical account of prominence and its interaction with boundaries in edge-marking languages is proposed, with implications for a cross-linguistic model of prosody. [Work supported by NSF.]

5aSC20. The kinematics of prominence in American English. Argyro Katsika (UCSB, 3432 University Dr., Santa Barbara, CA 93111, argyro@ucsb.edu), Jiyoung Jang (UCSB, Santa Barbara, CA), Louis Goldstein (USC, Los Angeles, CA), Jelena Krivokapic (Univ. of Michigan, Ann Arbor, MI), and Elliot Saltzman (BU, Boston, MA)

Speech units under prominence present longer, larger, and faster constriction gestures than their non-prominent counterparts. However, whether there are discrete degrees of prominence, and if so how many, has yet to be discovered, partly because the contribution of the information structure in marking prominence is unclear. Here, we employ electromagnetic articulography (EMA) to examine the effect of focus (unfocused, de-accented, broad focus, narrow focus, and contrastive focus) on the kinematics (position, duration, velocity, and stiffness) of the consonant gestures of the test words' stressed syllable. Test words varied in terms of length (one, two, three, and four syllables) and position of stress (first, second, or third syllable). Results from eight native speakers of American English indicate that prominent gestures become longer, larger, and faster in a way that consistently reflects three degrees of prominence. Nonetheless, kinematic dimensions differ in the number of degrees of prominence they distinguish, ranging from two levels (stiffness) to three (position and velocity) and four levels (duration). All dimensions distinguish contrastive focus from unfocused, while de-accented tends to be grouped with unfocused, and narrow focus with either contrastive or broad focus. A hierarchy of prominence is proposed taking into consideration categorical versus gradient distinctions. [Work supported by NSF.]

5aSC21. The scope of the kinematic effects of prominence in Greek. Argyro Katsika (UCSB, 3432 University Dr., Santa Barbara, CA 93111, argyro@ucsb.edu) and Karen Tsai (UCSB, Santa Barbara, CA)

It is well established that speech units under prominence present longer, larger, and faster constriction gestures than their non-prominent counterparts. However, little is known about the scope of these kinematic effects and how their scope interacts with prosodic boundaries. Previous research has mainly focused on the durational dimension (lengthening) and contrastive focus. The current study uses electromagnetic articulography (EMA) to investigate the scope of the kinematic effects of prominence related to non-contrastive focus as a function of stress position (word-initial, word-medial, word-final) and boundary type (phrase versus word boundary) in Greek. The kinematic dimensions examined are duration, position, velocity, and stiffness. Results from five speakers indicate that all kinematic dimensions affect the stressed syllable but also present both anticipatory and spillover effects. The strongest effect of the dimensions of duration and position is found within the stressed syllable, while the maximum of the velocity and stiffness effects is within the syllable that immediately follows the stressed one. Interestingly, when stress is final, the velocity and stiffness effects extend across prosodic boundaries, regardless of the boundary type, affecting the formation constriction of the first post-boundary consonant. A model of prominence is proposed within the framework of Articulatory Phonology. [Work supported by NIH.]

5aSC22. On the nature of working memory structures in phonological encoding. Matthew Masapollo (Graduate Program in Neurosci., Boston Univ., 677 Beacon St., Boston, MA 02215, mmasapol@bu.edu), Dante Smith (Graduate Program in Neurosci., Boston Univ., Boston, MA), and Frank Guenther (Dept. of Speech, Lang. and Hearing Sci. & Biomedical Eng., Boston Univ., Boston, MA)

Current model and theories of language and speech production commonly propose that speakers plan and execute sequences of phonemic segments by integrating and consolidating individual segments into cohesive memory structures or "chunks," which reduces processing load and improves motor performance. Yet, there is no consensus on the nature of the chunks buffered in memory during serial speech planning. To identify these structures, we investigated the generalization of motor chunking from training to transfer utterances. During training, subjects repeated *isolated* syllables containing non-native consonant clusters. Subjects produced these syllables with increased accuracy and speed after training, indicative of motor learning. After learning, we tested for generalization under higher memory load by having subjects repeat *pairs* of syllables that overlapped to varying degrees with the practiced syllables. We observed complete transfer of performance *speed* improvements to novel syllables containing previously practiced clusters, but only if they were practiced in the same syllable position (onset or coda). Practicing the whole syllable, however, resulted in greater *accuracy* improvements compared to practicing just the clusters, regardless of syllable position. Collectively, these findings suggest that working memory utilizes a syllabic structural frame with different representations for the same phonemes in different frame slots.

5aSC23. Automatic classification of French stops consonants. Clara Ponchard (UMR 7018 - CNRS/Sorbonne-Nouvelle, Laboratoire de Phoneä tique et Phonologie, 9 rue Jacques Dulud, Neuilly-sur-Seine 92200, France, cponchard@yahoo.fr), Didier Demolin (UMR 7018 - CNRS/Sorbonne-Nouvelle, Laboratoire de Phoneä tique et Phonologie, Paris, France), and Sergio Hassid (Hoä pital Erasme, Universiteä Libre de Bruxelles, Bruxelles, Belgium)

The purpose of this study is to analyze aerodynamic parameters involved in the production of French stops consonants. We are focusing our analysis on intraoral and subglottic pressure measurements. The main goal of this study is to automate the processing of aerodynamic data measurements in

order to contribute to provide reference data that are still limited in the literature. We are also analyzing pressure variations according to the voicing opposition, the intervocalic context, and different places of articulation. Phonetic analysis is carried out by means of statistical tests in order to analyze the descriptors considered relevant to understand the consonants' variations. Then, a supervised classification task to assess the relevance of these descriptors is made. Phonetic, automatic language processing and machine learning methods are processed in this work.

5aSC24. Fundamental frequency correlates with head movement evaluated at two contrasting speech production rates. Mark Tiede (Haskins Labs., 300 George St., Ste. 900, New Haven, CT 06511, tiede@haskins.yale.edu), Wei-Rong Chen (Haskins Labs., New Haven, CT), and D. H. Whalen (Speech-Language-Hearing Sci., CUNY, New York, NY)

The 720 phonetically balanced IEEE sentences have been recorded from eight speakers using electromagnetic articulometry (EMA) at "normal" and "fast" production rates. Participants self-selected their preferred normal rate and were instructed to produce the fast rate as quickly as possible without making errors (errorful productions were immediately repeated), resulting in an average 66.3% (0.4) duration reduction. EMA trajectories were recorded at 100 Hz with 44 100 Hz synchronized audio. The centroid of sensors attached to the upper incisors and the mastoid processes was used to characterize the 6DOF (X, Y, Z, roll, pitch, and yaw) deviation of the head in each sample from an initial occlusal plane reference position. Fundamental frequency, downsampled to 100 Hz, was correlated with this head movement signal using multiple linear regression. Mean adjusted R-squared values were uniformly higher for the faster production across all speakers: normal 0.77 (0.19) and fast 0.84 (0.18). Inclusion of jaw position as a regressor improved results to normal 0.89 (0.12) and fast 0.93 (0.11). The higher correlation at faster rates may be due to influence of observed increased head movement on laryngeal setting (and thus fundamental frequency) as production stress increases. These results have implications for the synthesis of head movement from acoustics and automatic modulation of synthetic voicing sources (electrolarynx). [Work supported by NIH.]

5aSC25. Coordination of tongue muscle shortening and volume shifting in speech. Maureen Stone (Univ. of Maryland Dental School, 650 W. Baltimore St. Rm 8207, Baltimore, MD 21201, mstone@umaryland.edu), Fangxu Xing (Massachusetts General Hospital, Boston, MA), Katie Garrett, Atefeh Boroun (Univ. of Maryland Dental School, Baltimore, MD), Jonghye Woo (Massachusetts General Hospital, Boston, MA), and Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD)

Tongue motion to execute speech motions uses complex 3-D muscle architecture. For example, the motion into /s/ requires the tongue to extend its tip anteriorly, groove its superior surface, and brace itself laterally. To facilitate this, orthogonal, interdigitated tongue muscles allow complex deformations and motions, which create subtle and local shape changes along the tongue surface. This talk will examine anterior-posterior tongue motion from two perspectives: muscle shortening and volume shifting. We measure tagged and cine MR images for two speech tasks, "a geese" versus "a souk," which move the tongue primarily forward versus backward. The talk will focus on the genioglossus, verticalis, transverse, superior, and inferior longitudinal muscles. These five muscles comprise over 50% of the tongue mass and deform it in three major directions: anterior-posterior, superior-inferior, and medial-lateral. The muscle shortening patterns will be compared to the percent of the tongue anterior to specific tooth landmarks for the consonant /s/ in both words. The /s/'s differ in word position and vowel context. Similarities between the two will help specify the coordination between muscles and deformation for /s/; differences will inform us as to the effect of vowel context/word placement.

5aSC26. Toward cross-speaker articulatory modeling. Asterios Toutios (Univ. of Southern California, 3740 McClintock Ave., EEB 400, Los Angeles, CA 90089, toutios@sipi.usc.edu), Reed Blaylock, Louis Goldstein, and Shrikanth S. Narayanan (Univ. of Southern California, Los Angeles, CA)

The recent increasing availability of comprehensive real-time MRI data of the vocal tract and concomitant progress in air-tissue boundary segmentation present novel opportunities for articulatory modeling. PCA-based articulatory models represent vocal tract configurations as weighted linear combinations of articulatory components that characterize vocal tract shaping patterns. Historically, most such models have been developed using data from a single speaker, and their direct application to data from multiple speakers may provide components that are not comparable across speakers. A technique that can address this issue is PARAFAC, which has been applied in the past to analyze data corresponding to tongue contour X-ray tracings of 10 English vowels from 5 speakers. PARAFAC introduces an additional level of weighting of the articulatory components that is constant for, and therefore characteristic of, each speaker. We revisited PARAFAC and ran a successful pilot on real-time MRI air tissue-boundaries of the entire vocal tract from 4 speakers, each uttering two repetitions of a set of 4 Shibboleth sentences, yielding 289 phones per speaker. Application on a much larger and diverse real-time MRI dataset already collected by our team will provide crucial progress toward true cross-speaker articulatory modeling. [Work supported by NIH and NSF.]

5aSC27. Intrinsic fundamental frequency of Amharic vowels. Maxine Van Doren (Linguist, Univ. of California San Diego, 9100 Regents Rd., Apt G, La Jolla, CA 92037, mevandoren@gmail.com)

Intrinsic fundamental frequency (IF0) refers to the cross-linguistic tendency for high vowels to have a higher F0 than low vowels (Whalen and Levitt, 1995). IF0 is often thought to be a result of mechanical coupling, where tongue movement causes changes in the laryngeal structures, perturbing the F0. Ohala and Eukel (1987) propose that high vowels lead to increased vertical tension on the true vocal folds, which raises F0. In contrast, the tongue-compression hypothesis proposes that low vowels induce vocal fold slackening, which lowers F0 (Ewan, 1979). The purpose of the present study is to investigate the cause of IF0 by comparing high, mid, and low vowels in Amharic, using audio and EGG signals to measure F0 and contact quotient for different vowel heights. Amharic was chosen because it contrasts three heights of central vowels. The goals are to confirm that IF0 effects are found in Amharic and determine if voice quality changes further support either hypothesis. Discussion will focus on how F0 and contact quotient differ by vowel height, and whether the results support the tongue-pull hypothesis (where IF0 is driven by high tongue position), the tongue-compression hypothesis (where IF0 is driven by low-back tongue position), or both.

5aSC28. Estimation of vocal fold geometry and stiffness from voice acoustics. Zhaoyan Zhang (Dept. of Head and Neck Surgery, UCLA, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Many speech applications require estimating vocal fold properties from the produced acoustics. While there have been many previous research on solving the inverse problem, they are often based on lumped-element models of phonation, whose model parameters are difficult to relate to realistic vocal fold properties. This study explores the feasibility of inferring physiologically realistic vocal fold properties, including vocal fold length, thickness, depth, anisotropic stiffness moduli, and the subglottal pressure, from the produced acoustics, using a three-dimensional phonation model and a Bayesian inference approach. To reduce the computational cost associated with the use of a three-dimensional model and the large number of inputs to

be estimated, we explore the possibility of improving computational efficiency and estimation accuracy using the large amount of three-dimensional simulation data available from our previous research. Preliminary results show that joint likelihood probabilities can be reasonably estimated based on the available simulation data, significantly reducing computational costs of Bayesian inference. The approach is able to estimate the control parameters of voice production from the produced acoustics with reasonable accuracy. It is observed that certain control parameters, particularly vocal fold stiffness, consistently have large estimation errors than others. [Work supported by NIH.]

5aSC29. Effect of medial surface shape on voice production in a MRI-based three-dimensional phonation model. Liang Wu (Dept. of Head and Neck Surgery, UCLA School of Medicine, 1000 Veteran Ave., Los Angeles, CA 90095, liangliang26@outlook.com) and Zhaoyan Zhang (Dept. of Head and Neck Surgery, UCLA School of Medicine, Los Angeles, CA)

Our previous studies using simplified vocal fold geometry have shown that medial surface shape, particularly the vertical thickness, has an important effect on the closure pattern of vocal fold vibration. The goal of this study is to investigate if similar effect can be observed for medial surface shape that occurs in realistic human phonation, using a parametric MRI-based three-dimensional vocal fold model. Manipulation of medial surface shape is achieved through a two-level control of the superior and inferior portion of the medial surface, based on experimental observations. Simulations show that, in general, both superior-medial bulging and inferior-medial bulging of the medial surface, which leads to an increased vertical thickness and a more rectangular glottal configuration, increase the closed quotient of the vocal fold vibration. Changes in medial surface shape also have significant effect on the phonation threshold pressure. This effect of the medial surface shape varies significantly across larynges, indicating the important effect of subject-specific laryngeal geometry. The results point to the importance of taking into consideration of the medial surface shape in clinical management of voice disorders. [Work supported by NIH.]

5aSC30. Analysis of genioglossus-centered tongue muscle function groups from dynamic magnetic resonance imaging. Fangxu Xing (Radiology, Massachusetts General Hospital, Rm. 304A, 55 Fruit St., Thiers Hall, Boston, MA 02114, fxing1@mgh.harvard.edu), Maureen Stone (Univ. of Maryland Dental School, Baltimore, MD), Tessa Goldsmith (Speech, Lang. and Swallowing Disord. & Reading Disabilities, Massachusetts General Hospital, Boston, MA), Jerry L. Prince (Johns Hopkins Univ., Baltimore, MD), Georges El Fakhri, and Jonghye Woo (Radiology, Massachusetts General Hospital, Boston, MA)

Internal tongue muscle functions vary in speech due to their anatomical structure differences and complex cooperative patterns. Assessment of muscle cooperative patterns and general mechanics provides insights into speech motor control and helps interpretation of clinical observations. In this work, we developed a method that revealed different function groups of all internal tongue muscles with respect to genioglossus (GG) during speech. Dynamic magnetic resonance imaging with tags was used to compute spatial and temporal muscle motion, which was deformed into a previously constructed statistical tongue atlas space. A pre-defined atlas muscle mask was used to extract motion at each muscle's location. We used GG as a major representation of the tongue's motion pattern and computed pattern correlation of each muscle's motion against GG over time. Muscle function groups were determined using the value of dynamic correlation series. The data from sixteen normal controls and three post-glossectomy patients were processed, revealing that mylohyoid, geniohyoid, and digastric muscles tended to function as a separate group while most other internal muscles followed GG. However, due to patients' post-glossectomy function loss, most of their internal muscles functioned as one unique group as a compensation strategy, providing information for further understanding of speech motor control.

5aSC31. An analysis of a thermoacoustic vocal tract model. Joshua Vawdrey (Phys., Utah Valley Univ., 10332 S Amaryllis St., Sandy, UT 84094, jvaw94@gmail.com), Stephen Beatty, Matthew King (Phys., Utah Valley Univ., Orem, UT), Masood Amin (Eng., Utah Valley Univ., Orem, UT), and Bonnie Andersen (Phys., Utah Valley Univ., Orem, UT)

A simplified human vocal tract has been modelled by Arai as an educational tool that is able to produce the formants of vowel sounds [T. Arai, *J. Acoust. Sci. Technol.* **27**, 6 (2006)]. By reconstructing Arai's model as a thermoacoustic engine with heat exchangers and a stack in one of the three tubes, heat produced sounds that are resonant with the model with no external driving mechanism needed. Two configurations of the stack placement in the three-tube model were tested, one with the stack in the constriction region and one with the stack located near the closed end. Varying the constriction location in the second configuration produces sounds with frequencies close to the male /e/ and /o/ first formant, as well as the male /ae/ and /a/ of the second formant. The first configuration with the stack in the constriction was not able to produce sound at the temperatures used in this study, likely due to radiation losses caused by its structure. Using a one-dimensional wave equation and applying boundary conditions at the ends of each of the three tubes results in a transcendental equation, which determines the formant frequencies as the back cavity length is increased.

5aSC32. Effects of vocal tract geometrical differences on flow and sound of sibilant fricatives. Tsukasa Yoshinaga (Toyohashi Univ. of Technol., 1-1 Hibarigaoka, Tempaku, Toyohashi, Aichi 560-8531, Japan, yoshinaga@me.tut.ac.jp), Kazunori Nozaki (Osaka Univ., Suita, Japan), Shigeo Wada (Osaka Univ., Toyonaka, Japan), and Akiyoshi Iida (Toyohashi Univ. of Technol., Toyohashi-shi, Aichi-ken, Japan)

Sibilant fricatives are known to be produced by controlling a turbulent jet flow in a front part of the vocal tract. However, the manner of the control in each individual is still unclear. In this study, the vocal tract geometries of sibilant fricatives /s/ and /ʃ/ were extracted from magnetic resonance imaging (MRI) and large eddy simulation was conducted to clarify the relationship among the vocal tract geometries, flows, and acoustic characteristics. Five male Japanese subjects assumed a supine position in the MR machine, and images of the vocal tract were corrected while the subject was sustaining /s/ and /ʃ/. From the corrected images, computational grids were constructed, and large eddy simulation of compressible flow was applied to simulate both the turbulent flow and sound generation. Results of the simulations showed that the configuration of jet flow in each vocal tract was different, and the maximum amplitude of sound source distributed in different positions in the front part of the vocal tracts. By extracting the characteristic dimensions of each vocal tract, we found that a space between lower incisor and tongue constriction is important for the formation of individual acoustic characteristics of sibilant fricatives.

5aSC33. Simulating tonic activations in speech production. Bryan Gick (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, B C V6T1Z4, Canada, gick@mail.ubc.ca) and Connor Mayer (Linguist, Univ. of California, Los Angeles, Los Angeles, CA)

Studies of speech production have often focused on transient events—those that happen over short temporal intervals. We know, however, that speech is made up of movements that can be distributed over longer durations as well (e.g., tongue bracing, oralization, articulatory setting, laryngeal state, harmony, etc.). Such events, which involve maintaining continuous activation of a particular muscle group tonically over a long duration, have

often been treated as qualitatively different from transient speech events. The present study considers examples of these types of movements in speech and non-speech (e.g., emotion expression, posture, etc.) domains. Biomechanical simulations are used to show how tonic activations operate on the same principles as transient ones (except for the difference in duration), and to show how tonic activations can overlap with multiple other activations—whether transient or tonic—through superposition [Bizzi *et al.* *Science* **253**, 5017 (1991)]. This work aims to show how these previously anomalous kinds of movements fit seamlessly into broader theories of movement and speech. [Funding from NIH Grant DC-002717 and NSERC RGPIN-2015-05099.]

5aSC34. A real-time system for visualizing tongue position during vowel phonation. Bernard Wang (Ctr. for Comput. Res. in Music and Acoust., Stanford, 160 Comstock Circle, Stanford, CA 94305, bernardcwang@gmail.com) and Julius O. Smith (Ctr. for Comput. Res. in Music and Acoust., Stanford, Stanford, CA)

A real-time system to estimate and visualize the position of the tongue during vowel phonation is presented. The system uses Linear Predictive Coding (LPC) to track formants F1–F3 of input speech frames. An algorithm proposed by Ladefoged *et al*. is used to map formants F1–F3 to tongue position [*J. Acoust. Soc. Am.* **64**, 1027 (1978)]. 2-D visualizations of the tongue are created with cubic splines, which are rendered each input frame to create a real-time animation. The proposed system could serve as a pedagogical tool for language learners and singers by providing visual feedback of the inner mechanics of the vocal tract during vowel phonation.

5aSC35. Learning a representation of tongue dynamics from unlabeled ultrasound videos. Hongcui Wang (Institut Langevin, ESPCI Paris, PSL Univ., CNRS, Sorbonne Université, 1 rue Jussieu, Paris 75005, France, laurelwind@gmail.com), Pierre Roussel (I), and Bruce Denby (Institut Langevin, ESPCI Paris, PSL Univ., CNRS, Sorbonne Université, Paris, France)

Ultrasound imaging of the tongue has been used for decades in studies of speech production and speech motor control, for silent speech interfaces, and in numerous other areas. Despite substantial efforts, however, extraction of reliable features from ultrasound tongue data remains a challenge due to speckle noise and acoustic propagation issues. Recently, Representation Learning has emerged in a variety of fields as a powerful means of generating useful representations of underlying structure in raw, high-dimensional data. In its unsupervised form, Representation Learning discovers structures in unlabelled data, thereby eliminating the need for a time-consuming labelling step. The present work is believed to be the first use of unsupervised Representation Learning to reveal structures related to tongue dynamics in unlabelled ultrasound video. A 3-D Convolutional Neural Network examining a series of unlabelled 60 Hz tongue images is found to accurately predict unseen future images even for large interframe tongue displacements. By comparing the 3DCNN prediction error to that of a simple previous-frame predictor, tongue trajectories containing transitions between regions of acoustic stability can be identified and correlated with formant trajectories in a spectrogram. Prospects for leveraging the tongue dynamic representation for use in subsequent speech processing tasks will be discussed.

Session 5aUW

Underwater Acoustics: Remote Sensing, Inversion, and Passive Sensing

Emmanuel Skarsoulis, Chair
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Contributed Papers

8:15

5aUW1. Compressive two-dimensional beamforming for localization of tip vortex cavitation. Yongsung Park (Scripps Inst. of Oceanogr., Seoul National University, Bldg 36 - Rm. 212, Gwanak-ro 1, Gwanak-gu, Seoul 08826, South Korea, ysparkwin@snu.ac.kr), Peter Gerstoft (Scripps Inst. of Oceanogr., La Jolla, CA), and Woojae Seong (Seoul National Univ., Seoul, South Korea)

Noises induced by propeller tip vortex cavitation (TVC) have sparse sources near the propeller tip. Localization of these noise sources with sensor arrays involves the direction-of-arrival (DOA) estimation. Provided the noise sources are sparsely located, compressive sensing (CS) is used for two-dimensional (2-D) DOA estimation, which estimates both azimuth and elevation angles of the sources. With a planar array configuration, the 2-D DOA estimation problem is formulated in the CS framework and CS has shown less ambiguity and higher resolution capabilities compared to traditional DOA estimation schemes even under a single snapshot data. The superior performance of CS is demonstrated on experimental data from cavitation tunnel experiments. Two experimental data are considered. For the first experiment with one known source location, the true DOA has been estimated using CS. The second case involves initially intended TVC noises, and CS has estimated DOAs near the upper downstream area of the propeller, where TVC noises are generally generated.

8:30

5aUW2. A two-way approach to adapt reduced-scale laboratory experiments and corresponding numerical simulations of offshore seismic surveys in complex marine environments. Nathalie Favretto-Cristini, Bence Solymosi (Aix-Marseille Université, CNRS, Centrale Marseille, LMA, Marseille, France), Paul Cristini (Aix-Marseille Université, CNRS, Centrale Marseille, LMA, 4, Impasse Nikola Tesla, CS40006, Marseille 13013, France, cristini@lma.cnrs-mrs.fr), Vadim Monteiller (Aix-Marseille Université, CNRS, Centrale Marseille, LMA, Marseille, France), Bjorn Ursin (Norwegian Univ. of Sci. and Technol., Trondheim, Norway), and Dimitri Komatitsch (Aix-Marseille Université, CNRS, Centrale Marseille, LMA, Marseille, France)

Recently, laboratory experiments have been reintroduced in the ideas-to-applications pipeline for geophysical issues. Benefiting from recent technological advances, we believe that in the coming years, laboratory experiments can play a major role, in support of field experiments and numerical modeling, to explore some of the current challenges of seismic or acoustic imaging in terms of, for instance, acquisition design or benchmarking of new imaging techniques at a low cost. But having confidence in the quality and the accuracy of the experimental data obtained in a complex configuration that mimics at a reduced scale a real geological environment is an essential prerequisite. This requires a robust framework regardless of the configuration studied. The goal of this work is to provide a global reflection on this framework in the context of offshore seismics. To illustrate this framework, we rely on a reduced-scale model representing a 3-D complex-shaped salt body buried in sedimentary layers with curved surfaces. Zero-offset and offset reflection data are collected on this model in a water tank using a conventional pulse-echo technique. We follow a cross-validation approach that allows, through the comparison between the experimental data and the numerical simulation of wave propagation, to point out both the improvements of the experimental setup that must still be made to increase the accuracy of the experiments and the limitations of the numerical tools that must be tackled.

8:45

5aUW3. Characterization of internal waves in SAS imagery via ray tracing. David J. Pate (Georgia Tech Res. Inst., 7220 Richardson Rd., Smyrna, GA 30080, david.pate@gtri.gatech.edu), Daniel Cook (Georgia Tech Res. Inst., Smyrna, GA), Anthony P. Lyons (Univ. of New Hampshire, Durham, NH), and Roy E. Hansen (Norwegian Defence Res. Establishment (FFI), Kjeller, Norway)

Synthetic aperture sonar imagery often captures features that appear similar to sand waves but are actually pockets of denser water traveling as isolated waves along the seafloor. These pockets of cold water refract acoustic waves like a lens, causing intensity peaks and shadows that resemble medium to large scale sand waves. This work uses dynamic ray tracing to predict the intensity return as affected by refraction. First, we explore the nature of the intensity pattern created by internal waves of various shapes and sizes. Then, we use an optimization-based approach to solve the inverse problem: given an intensity pattern, determine the size, shape, and location of the internal wave that created it.

9:00

5aUW4. Investigation of bathymetry problem using kinetic model. Elizaveta R. Liu (Dept. of Information Technologies, Mathematical and Comput. Modelling, Far Eastern Federal Univ., Sukhanova 8, Vladivostok, Primorskii krai 690090, Russian Federation, elizavetarobertovna@gmail.com), Andrei Sushchenko (Dept. of Information Technologies, Mathematical and Comput. Modelling, Far Eastern Federal Univ., Vladivostok, Primorskii krai, Russian Federation), and Igor Prokhorov (Inst. of Appl. Mathematics FEB RAS, Vladivostok, Primorskii krai, Russian Federation)

In this paper, we investigate a bathymetry problem, the seabottom surface reconstruction, using side scan sonars. A sonar emits pulses of sounds and detects echoes. The sound waves bounce off the surface, and the sonar receives the backscattered signal. Gauging water depth, using acoustic technology, involves measuring the time taken for sound waves to travel between the vessel and the seabottom and back again. To describe sound propagation in a fluctuating ocean, a mathematical model, based on the radiative transfer equation, is used. The solution of the direct problem, consisting of determining the wave energy flux density, has been obtained in the single and double scattering approximations. Further, we consider inverse problem which involves the determination of the bathymetric function, which describes the seabottom level variation from the average horizontal plane. As a solution to the problem, a non-linear differential equation has been deduced under assumptions for the directivity pattern of the receiving antenna. An adaptive numerical method is developed for the analysis of the

bathymetry problem. Computational experiments, showing the influence of a double scattering signal on the seabottom reconstruction, are conducted.

9:15

5aUW5. Broadband horizontal wavenumber estimation using block sparse Bayesian learning. Haiqiang Niu (Inst. of Acoust., Chinese Acad. of Sci., No. 21 North 4th Ring Rd., Beijing 100190, China, nhq@mail.ioa.ac.cn), Peter Gerstoft, Emma Reeves Ozanich (Scripps Inst. of Oceanogr., La Jolla, CA), Zhenglin Li, and ZaiXiao Gong (Inst. of Acoust., Chinese Acad. of Sci., Beijing, China)

The horizontal wavenumbers are estimated by block sparse Bayesian learning for broadband signals received by a vertical line array in ocean waveguides. The dictionary matrix consists of multi-frequency modal depth functions derived from WKB approximation, which are associated with horizontal wavenumbers. The dispersion relation for multi-frequency horizontal wavenumbers is also taken into account to generate the dictionary. With the constraint of block sparsity, the sparse Bayesian learning approach is shown to extract the horizontal wavenumbers and corresponding modal depth functions with a high precision, while the prior of sea bottom information, moving source, and source locations is not needed. The performance is demonstrated by simulations and experimental data.

9:30

5aUW6. Underwater terrain matching navigation based on Gaussian process regression with a multi-beam Bathymetric sonar. Dongdong Peng (College of Underwater Acoust. Eng., Harbin Eng. Univ., No.145 Nantong St., Nangang Dist., Harbin, Heilongjiang 150001, China, pengdongdong@hrbeu.edu.cn), Jiaqi Gao, and Tian Zhou (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China)

Terrain matching navigation estimates the position of underwater vehicle by matching measured terrain against a prior map, which is attractive to fix the drift inherent to inertial navigation system. With a multi-beam bathymetric sonar, a local terrain is usually reconstructed by deterministic interpolation methods to match the prior map, such as linear and spline interpolation methods. However, these deterministic method change the statistical properties of the terrain, which reduce the positioning accuracy. To improve the positioning accuracy, a probabilistic interpolation method based on Gaussian process regression is proposed to reconstruct the terrain in this paper. Different from the deterministic interpolation, Gaussian process regression can not only maintain the statistical properties as far as possible but also give the uncertainty of a interpolated depth. The uncertainty can then be fused into the measurement error in the maximum likelihood estimation method to improve the positioning accuracy. Simulation experiments in a real underwater map demonstrate that the proposed method is feasible and more accurate than the traditional deterministic interpolation for underwater terrain matching navigation.

9:45–10:00 Break

10:00

5aUW7. Source kernels for noise cross-correlation. 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 (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

A wave-theoretic tool, the source kernel, is introduced to describe the sensitivity of the finite-frequency underwater noise cross-correlation function to the location of the noise sources taking into account the refractive features of the ocean environment. The cross correlation of the noise field at two receivers conveys information about the corresponding time-domain Green's function (TDGF), provided that sufficient energy is channeled into the acoustic paths connecting the two locations. The efficiency of the TDGF recovery is determined by the receiver and noise source locations and characteristics, as well as by the refraction properties of the ocean sound channel. The source kernel takes these characteristics into account and highlights the locations where noise sources can have maximum impact on the cross-correlation output. The distribution and coherence of the noise sources can be specified in a variety of ways to investigate a wide range of scenarios, from random noise-source distributions to coherent radiators. [Work supported by the Office of Naval Research.]

10:15

5aUW8. Simulation research of underwater passive direction finding technology. Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn) and Kaiyu Tang (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang, China)

As one of the indispensable technologies for underwater target detecting, underwater acoustic passive detection technology has always been a hot spot in marine system research. A vector sensor can detect underwater targets by combination with the pressure and vibration velocity information; vector sensor array can obtain higher gain than pressure array with the same number hydrophones. Vector sensor can reduce the length of the underwater sensor array and can detect underwater targets in the case of low signal to noise ratio. This is the goal that many researchers strive to solve for many years; therefore, vector sensor passive detection technology has far-reaching scientific research value.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, and so it is of great significance in the research of underwater acoustic detection technology.

10:30

5aUW9. Improving passive acoustic monitoring applications to the endangered Cook inlet beluga whale. Ming Zhong (Microsoft, One Microsoft Way, Redmond, WA, mizhong@microsoft.com), Manuel Castellote (National Oceanic and Atmospheric Administration, Seattle, WA), Rahul Dodhia, Juan Lavista Ferres (Microsoft, Redmond, WA), Mandy Keogh (Alaska Dept. of Fish and Game, Juneau, AK), and Arial Brewer (National Oceanic and Atmospheric Administration, Seattle, WA)

A decade after the Cook Inlet beluga was listed as endangered in 2008, its population has shown no signs of recovery. Lack of ecology knowledge limits our understanding of, and ability to manage, potential threats impeding recovery of this declining population. NOAA Fisheries, in partnership with the Alaska Department of Fish and Game, initiated a passive acoustics program in 2017 to monitor beluga seasonal occurrence by deploying a series of acoustic moorings, followed by months' work of manual validation for the detectors' output. To reduce this labor intensive and time-consuming process, we extracted a series of spectrograms from the sound files containing validated detections, and built 4 deep learning convolutional neural networks (CNN) with fine tuning of parameters. The final model is an ensemble of these individually optimized models, and achieves 96.57% precision and 92.26% recall on testing data. As a comparison, current detectors tend to trigger more false positives, which result in 20%–60% precision when human noise is present and 85%–95% precision in quiet areas. Following the success of the model with its easy generalizations to other acoustic detection problems, our next step will be comparing results from non-validated raw data to adopt this updated analysis process.

10:45

5aUW10. Analysis for approach of passive measurement based on waveguide invariant. Juan Hui (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang 150001, China, huijuan@hrbeu.edu.cn) and Kaiyu Tang (Underwater Acoust., Harbin Eng. Univ., Harbin, Heilongjiang, China)

Nowadays, the development of underwater acoustic positioning and measuring direction technology becomes more and more important. A method which belongs to a passive direction based on waveguide invariant, could be used to measure a heading angle of a target source with guide source technology, combined with a single acoustic vector hydrophone at a low frequency in the epicontinental. At first, in the epicontinental ocean, at

a low frequency, simulating the channel of ocean according to normal mode theory using the software called KrakenC. Then, the vector acoustic pressure field could be calculated, getting the schematic diagram about a stable interference structure of epicontinental ocean in a low frequency. Waveguide invariant could describe the situation about that. To pick up, waveguide invariant becomes the most important part of this paper. This process get with the movement of the guide source and the target source; also, an image processing method called Hough transform is essential. Finally, waveguide invariant is acquired. At the same time, by that, the heading angle of the target source is easy to be estimated. When detailed information of the ocean have not acquired enough, this method still works well, and simulation data for the experiment prove the feasibility of the model.

11:00

5aUW11. The deconvolved conventional beamforming for conformal array. Dajun Sun (Harbin Eng. Univ., Harbin, China), Chao Ma (College of Underwater Acoust. Eng., Harbin Eng. Univ., Nantong Str, No. 145, Nangang Dist, Harbin 150001, China, machao93@hrbeu.edu.cn), and Jidan Mei (Harbin Eng. Univ., Harbin, China)

As the conformal array has the advantages over the large array aperture, superior aeroacoustic dynamics, and wide angle cover range, it is widely used in radar and sonar system. The deconvolved conventional beamforming (dCv) can yield not only the improved direction of arrival (DOA) estimation accuracy in terms of a narrower beamwidth and suppressed side lobes but also the higher directivity index than the CBF. This paper uses an extended R-L deconvolution algorithm and applies the dCv in conformal array. Simulation and experiment results show that the extended R-L can be applied to an arbitrary conformal array with a shift-variant point spread function (PSF). Beyond the high-resolution and high array gain performance advantages, the extended R-L algorithm is shown to be suitable for coherent signals. The results also show that it is better to use the actual PSF than the theoretical PSF when there exit position errors.

ETHICAL PRINCIPLES OF THE ACOUSTICAL SOCIETY OF AMERICA FOR RESEARCH INVOLVING HUMAN AND NON-HUMAN ANIMALS IN RESEARCH AND PUBLISHING AND PRESENTATIONS

The Acoustical Society of America (ASA) has endorsed the following ethical principles associated with the use of human and non-human animals in research, and for publishing and presentations. The principles endorsed by the Society follow the form of those adopted by the American Psychological Association (APA), along with excerpts borrowed from the Council for International Organizations of Medical Sciences (CIOMS). The ASA acknowledges the difficulty in making ethical judgments, but the ASA wishes to set minimum socially accepted ethical standards for publishing in its journals and presenting at its meetings. These Ethical Principles are based on the principle that the individual author or presenter bears the responsibility for the ethical conduct of their research and is publication or presentation.

Authors of manuscripts submitted for publication in a journal of the Acoustical Society of America or presenting a paper at a meeting of the Society are obligated to follow the ethical principles of the Society. Failure to accept the ethical principles of the ASA shall result in the immediate rejection of manuscripts and/or proposals for publication or presentation. False indications of having followed the Ethical Principles of the ASA may be brought to the Ethics and Grievances Committee of the ASA.

APPROVAL BY APPROPRIATE GOVERNING AUTHORITY

The ASA requires all authors to abide by the principles of ethical research as a prerequisite for participation in Society-wide activities (e.g., publication of papers, presentations at meetings, etc.). Furthermore, the Society endorses the view that all research involving human and non-human vertebrate animals requires approval by the appropriate governing authority (e.g., institutional review board [IRB], or institutional animal care and use committee [IACUC], Health Insurance Portability and Accountability Act [HIPAA], or by other governing authorities used in many countries) and adopts the requirement that all research must be conducted in accordance with an approved research protocol as a precondition for participation in ASA programs. If no such governing authority exists, then the intent of the ASA Ethical Principles described in this document must be met. All research involving the use of human or non-human animals must have met the ASA Ethical Principles prior to the materials being submitted to the ASA for publication or presentation.

USE OF HUMAN SUBJECTS IN RESEARCH-Applicable when human subjects are used in the research

Research involving the use of human subjects should have been approved by an existing appropriate governing authority (e.g., an institutional review board [IRB]) whose policies are consistent with the Ethical Principles of the ASA **or** the research should have met the following criteria:

Informed Consent

When obtaining informed consent from prospective participants in a research protocol that has been approved by the appropriate and responsible-governing body, authors must have clearly and simply specified to the participants beforehand:

1. The purpose of the research, the expected duration of the study, and all procedures that were to be used.
2. The right of participants to decline to participate and to withdraw from the research in question after participation began.
3. The foreseeable consequences of declining or withdrawing from a study.
4. Anticipated factors that may have influenced a prospective participant's willingness to participate in a research project, such as potential risks, discomfort, or adverse effects.
5. All prospective research benefits.
6. The limits of confidentially.
7. Incentives for participation.
8. Whom to contact for questions about the research and the rights of research participants. The office/person must have willingly provided an atmosphere in which prospective participants were able to ask questions and receive answers.

Authors conducting intervention research involving the use of experimental treatments must have clarified, for each prospective participant, the following issues at the outset of the research:

1. The experimental nature of the treatment;
2. The services that were or were not to be available to the control group(s) if appropriate;
3. The means by which assignment to treatment and control groups were made;
4. Available treatment alternatives if an individual did not wish to participate in the research or wished to withdraw once a study had begun; and
5. Compensation for expenses incurred as a result of participating in a study including, if appropriate, whether reimbursement from the participant or a third-party payer was sought.

Informed Consent for Recording Voices and Images in Research

Authors must have obtained informed consent from research participants prior to recording their voices or images for data collection unless:

1. The research consisted solely of naturalistic observations in public places, and it was not anticipated that the recording would be used in a manner that could have caused personal identification or harm, or
2. The research design included deception. If deceptive tactics were a necessary component of the research design, consent for the use of recordings was obtained during the debriefing session.

Client/Patient, Student, and Subordinate Research Participants

When authors conduct research with clients/patients, students, or subordinates as participants, they must have taken steps to protect the prospective participants from adverse consequences of declining or withdrawing from participation.

Dispensing With Informed Consent for Research

Authors may have dispensed with the requirement to obtain informed consent when:

1. It was reasonable to assume that the research protocol in question did not create distress or harm to the participant and involves:
   a. The study of normal educational practices, curricula, or classroom management methods that were conducted in educational settings
   b. Anonymous questionnaires, naturalistic observations, or archival research for which disclosure of responses would not place participants at risk of criminal or civil liability or damage their financial standing, employability, or reputation, and confidentiality
   c. The study of factors related to job or organization effectiveness conducted in organizational settings for which there was no risk to participants' employability, and confidentiality.
2. Dispensation is permitted by law.
3. The research involved the collection or study of existing data, documents, records, pathological specimens, or diagnostic specimens, if these sources are publicly available or if the information is recorded by the investigator in such a manner that subjects cannot be identified, directly or through identifiers linked to the subjects.

Offering Inducements for Research Participation

(a) Authors must not have made excessive or inappropriate financial or other inducements for research participation when such inducements are likely to coerce participation.

(b) When offering professional services as an inducement for research participation, authors must have clarified the nature of the services, as well as the risks, obligations, and limitations.

Deception in Research

(a) Authors must not have conducted a study involving deception unless they had determined that the use of deceptive techniques was justified by the study's significant prospective scientific, educational, or applied value and that effective non-deceptive alternative procedures were not feasible.

(b) Authors must not have deceived prospective participants about research that is reasonably expected to cause physical pain or severe emotional distress.

(c) Authors must have explained any deception that was an integral feature of the design and conduct of an experiment to participants as early as was feasible, preferably at the conclusion of their participation, but no later than at the conclusion of the data collection period, and participants were freely permitted to withdraw their data.

Debriefing

(a) Authors must have provided a prompt opportunity for participants to obtain appropriate information about the nature, results, and conclusions of the research project for which they were a part, and they must have taken reasonable steps to correct any misconceptions that participants may have had of which the experimenters were aware.

(b) If scientific or humane values justified delaying or withholding relevant information, authors must have taken reasonable measures to reduce the risk of harm.

(c) If authors were aware that research procedures had harmed a participant, they must have taken reasonable steps to have minimized the harm.

HUMANE CARE AND USE OF NON-HUMAN VERTEBRATE ANIMALS IN RESEARCH-Applicable when non-human vertebrate animals are used in the research

The advancement of science and the development of improved means to protect the health and well being both of human and non-human vertebrate animals often require the use of intact individuals representing a wide variety of species in experiments designed to address reasonable scientific questions. Vertebrate animal experiments should have been undertaken only after due consideration of the relevance for health, conservation, and the advancement of scientific knowledge. (Modified from the Council for International Organizations of Medical Science (CIOMS) document: "International Guiding Principles for Biomedical Research Involving Animals1985"). Research involving the use of vertebrate animals should have been approved by an existing appropriate governing authority (e.g., an institutional animal care and use committee [IACUC]) whose policies are consistent with the Ethical Principles of the ASA **or** the research should have met the following criteria:

The proper and humane treatment of vertebrate animals in research demands that investigators:

1. Acquired, cared for, used, interacted with, observed, and disposed of animals in compliance with all current federal, state, and local laws and regulations, and with professional standards.
2. Are knowledgeable of applicable research methods and are experienced in the care of laboratory animals, supervised all procedures involving animals, and assumed responsibility for the comfort, health, and humane treatment of experimental animals under all circumstances.
3. Have insured that the current research is not repetitive of previously published work.
4. Should have used alternatives (e.g., mathematical models, computer simulations, etc.) when possible and reasonable.
5. Must have performed surgical procedures that were under appropriate anesthesia and followed techniques that avoided infection and minimized pain during and after surgery.
6. Have ensured that all subordinates who use animals as a part of their employment or education received instruction in research methods and in the care, maintenance, and handling of the species that were used, commensurate with the nature of their role as a member of the research team.
7. Must have made all reasonable efforts to minimize the number of vertebrate animals used, the discomfort, the illness, and the pain of all animal subjects.
8. Must have made all reasonable efforts to minimize any harm to the environment necessary for the safety and well being of animals that were observed or may have been affective as part of a research study.
9. Must have made all reasonable efforts to have monitored and then mitigated any possible adverse affects to animals that were observed as a function of the experimental protocol.
10. Who have used a procedure subjecting animals to pain, stress, or privation may have done so only when an alternative procedure was unavailable; the goal was justified by its prospective scientific, educational, or applied value; and the protocol had been approved by an appropriate review board.
11. Proceeded rapidly to humanely terminate an animal's life when it was necessary and appropriate, always minimizing pain and always in accordance with accepted procedures as determined by an appropriate review board.

PUBLICATION and PRESENTATION ETHICS-For publications in ASA journals and presentations at ASA sponsored meetings

Plagiarism

Authors must not have presented portions of another's work or data as their own under any circumstances.

Publication Credit

Authors have taken responsibility and credit, including authorship credit, only for work they have actually performed or to which they have substantially contributed. Principal authorship and other publication credits accurately reflect the relative scientific or professional contributions of the individuals involved, regardless of their relative status. Mere possession of an institutional position, such as a department chair, does not justify authorship credit. Minor contributions to the research or to the writing of the paper should have been acknowledged appropriately, such as in footnotes or in an introductory statement.

Duplicate Publication of Data

Authors did not publish, as original data, findings that have been previously published. This does not preclude the republication of data when they are accompanied by proper acknowledgment as defined by the publication policies of the ASA.

Reporting Research Results

If authors discover significant errors in published data, reasonable steps must be made in as timely a manner as possible to rectify such errors. Errors can be rectified by a correction, retraction, erratum, or other appropriate publication means.

DISCLOSURE OF CONFLICTS OF INTEREST

If the publication or presentation of the work could directly benefit the author(s), especially financially, then the author(s) must disclose the nature of the conflict:

1) The complete affiliation(s) of each author and sources of funding for the published or presented research should be clearly described in the paper or publication abstract.

2) If the publication or presentation of the research would directly lead to the financial gain of the authors(s), then a statement to this effect must appear in the acknowledgment section of the paper or presentation abstract or in a footnote of a paper.

3) If the research that is to be published or presented is in a controversial area and the publication or presentation presents only one view in regard to the controversy, then the existence of the controversy and this view must be provided in the acknowledgment section of the paper or presentation abstract or in a footnote of a paper. It is the responsibility of the author to determine if the paper or presentation is in a controversial area and if the person is expressing a singular view regarding the controversy.

Sustaining Members of the Acoustical Society of America

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OF AMERICA

The Acoustical Society is grateful for the financial assistance being given by the Sustaining Members listed below and invites applications for sustaining membership from other individuals or corporations who are interested in the welfare of the Society.

Application for membership may be made to the Executive Director of the Society and is subject to the approval of the Executive Council. Dues of $1000.00 for small businesses (annual gross below $100 million) and $2000.00 for large businesses (annual gross above $100 million or staff of commensurate size) include a subscription to the *Journal* as well as a yearly membership certificate suitable for framing. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year.

Additional information and application forms may be obtained from Elaine Moran, Office Manager, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300. Telephone: (516) 576-2360; E-mail: elaine@acousticalsociety.org

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www.ncac.com
Indianapolis, Indiana
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National Gypsum Company
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Wenger Corporation
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Owatonna, Minnesota
Design and Manufacturing of Architectural
Acoustical Products including Absorbers, Diffusers, Modular Sound
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The Wyle Acoustics Group provides a wide range of professional services focused on acoustics, vibration, and their allied technologies, including services to the aviation industry

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APPLICATION FOR SUSTAINING MEMBERSHIP

The Bylaws provide that any person, corporation, or organization contributing annual dues as fixed by the Executive Council shall be eligible for election to Sustaining Membership in the Society.

Dues have been fixed by the Executive Council as follows: $1000 for small businesses (annual gross below $100 million); $2000 for large businesses (annual gross above $100 million or staff of commensurate size). Dues include one year subscription to *The Journal of the Acoustical Society of America* and programs of Meetings of the Society. Please do not send dues with application. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year. If elected, you will be billed.

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It is understood that a Sustaining Member will not use the membership for promotional purposes.

Signature of company representatives making application:

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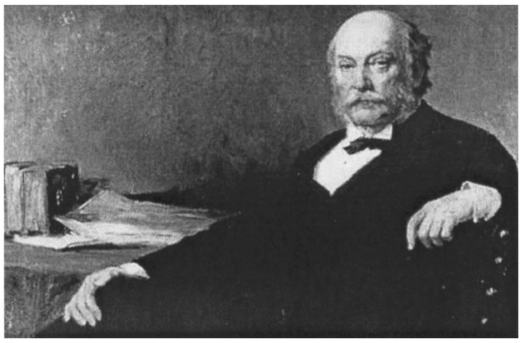
Please send completed applications to: Executive Director, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, (516) 576-2360, asa@acousticalsociety.org



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Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of the application; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

BENEFITS OF MEMBERSHIP	full Member	Associate	ce-Associate	Student
JASA Online–Vol. 1 (1929) to present	*	*	*	*
JASA tables of contents e-mail alerts	*	*	*	*
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Acoustics Today–the quarterly magazine	*	*	*	*
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Acoustics Research Letters Online (ARLO)–online archive	*	*	*	*
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Meeting Calls for Papers	Online	Online	Online	Online
Reduced Meeting Registration Fees	*	*		*
Society Membership Directory	Online	Online	Online	Online
Electronic Announcements	*	*	*	*
Physics Today	*	*	*	*
Eligibility to vote and hold office in ASA	*			
Eligibility to be elected Fellow	*	*		
Participation in ASA Committees	*	*	*	*

QUALIFICATIONS FOR EACH GRADE OF MEMBERSHIP AND ANNUAL DUES

Student: Any student interested in acoustics who is enrolled in an accredited college or university for half time or more (at least eight semester hours). Dues: $50 per year.
Associate: Any individual interested in acoustics. Dues: $115 per year. After five years, the dues of an Associate increase to that of a full Member.
Corresponding Electronic Associate: Any individual residing in a developing country who wishes to have access to ASA's online publications only including *The Journal of the Acoustical Society of America* and Meeting Programs [see http://acousticalsociety.org/membership/membership_and_benefits]. Dues $50 per year.
Member: Any person active in acoustics, who has an academic degree in acoustics or in a closely related field or who has had the equivalent of an academic degree in scientific or professional experience in acoustics, shall be eligible for election to Membership in the Society. A nonmember applying for full Membership will automatically be made an interim Associate Member, and must submit $115 with the application for the first year's dues. Election to full Membership may require six months or more for processing; dues as a full Member will be billed for subsequent years.

JOURNAL OPTIONS AND COSTS FOR FULL MEMBERS AND ASSOCIATE MEMBERS ONLY

- ONLINE JOURNAL. All members will receive access to the *The Journal of the Acoustical Society of America (JASA)* at no charge in addition to dues.
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Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of this form; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

PART I. TO BE COMPLETED BY ALL APPLICANTS (Please print or type all entries)

CHECK ONE BOX IN EACH COLUMN ON THE RIGHT	☐ NON-MEMBER APPLYING FOR: ☐ MEMBER REQUESTING TRANSFER TO:	☐ STUDENT MEMBERSHIP ☐ ASSOCIATE MEMBERSHIP ☐ CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP ☐ FULL MEMBERSHIP	Note that your choice of journal option *may* increase or decrease the amount you must remit.

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CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
NAME OF ORGANIZATION OR BUSINESS			
DEPARTMENT			
ORGANIZATION ADDRESS (STREET & NUMBER)			
CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
BUSINESS TELEPHONE: AREA CODE/NUMBER	FAX: AREA CODE/NUMBER	HOME TELEPHONE: AREA CODE/NUMBER	
E-MAIL ADDRESS: (PRINT CLEARLY)	MOBILE PHONE: AREA CODE/NUMBER		
DATE AND PLACE OF BIRTH (Req'd for Awards and Emeritus Status)	SEX: ☐ Female ☐ Male		
HIGHEST ACADEMIC DEGREE	DATE OF DEGREE	FIELD	INSTITUTION GRANTING DEGREE
OTHER DEGREE	MONTH/YEAR	FIELD	INSTITUTION GRANTING DEGREE

CHECK PERFERRED ADDRESS FOR MAIL: ☐ HOME ☐ ORGANIZATION

Part I Continued ➛

PART I CONTINUED: ACOUSTICAL AREAS OF INTEREST TO APPLICANT. Indicate your three main areas of interest below, using 1 for your main interest, 2 for your second, and 3 for your third interest. (DO NOT USE CHECK MARKS.)

- ☐ ACOUSTICAL OCEANOGRAPHY **M**
- ☐ ANIMAL BIOACOUSTICS **L**
- ☐ ARCHITECTURAL ACOUSTICS **A**
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- ☐ NOISE & NOISE CONTROL **D**
- ☐ PHYSICAL ACOUSTICS **E**
- ☐ PSYCHOLOGICAL & PHYSIOLOGICAL ACOUSTICS **F**
- ☐ SIGNAL PROCESSING IN ACOUSTICS **N**
- ☐ SPEECH COMMUNICATION **H**
- ☐ STRUCTURAL ACOUSTICS & VIBRATION **G**
- ☐ UNDERWATER ACOUSTICS **J**

PART II: APPLICATION FOR STUDENT MEMBERSHIP

NAME AND ADDRESS OF COLLEGE OR UNIVERSITY WHERE PRESENTLY ENROLLED		
DEGREE EXPECTED	MONTH & YEAR DEGREE EXPECTED	NUMBER OF SEMESTER HOURS ATTENDED THIS SEMESTER
PRINT NAMES & E-MAIL ADDRESSES OF TWO FACULTY MEMBERS CERTIFYING THAT YOU ARE REGISTERED FOR AT LEAST ONE-HALF OF FULL TIME		
SIGNATURES OF THE TWO FACULTY MEMBERS LISTED ABOVE CERTIFYING THAT YOU ARE REGISTERED AT LEAST HALF TIME		
SIGNATURE OF APPLICANT		DATE

PART III: APPLICATION FOR ASSOCIATE MEMBERSHIP, CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP OR FULL MEMBERSHIP (and interim Associate Membership)

SUMMARIZE YOUR MAJOR PROFESSIONAL EXPERIENCE on the lines below: list employers, duties and position titles, and dates, beginning with your present position. Attach additional sheets if more space is required.

CONTRIBUTIONS TO ACOUSTICS: LIST MAIN PUBLICATIONS, PATENTS, ETC. Attach separate sheets if required.

SPONSORS AND REFERENCES: An application for full Membership requires the names, and email addresses of two references who must be **full Members or Fellows** of the Acoustical Society. Names and signatures are NOT required for Associate Membership, Corresponding Electronic Associate Membership or Student Membership applications.

PRINT NAME OF REFERENCE (required for Full Member applications only)	PRINT NAME OF SECOND REFERENCE (required for Full Member applications only)
EMAIL ADDRESS OF REFERENCE	EMAIL ADDRESS OF SECOND REFERENCE
SIGNATURE OF APPLICANT	DATE

MAIL THIS COMPLETED APPLICATION, WITH APPROPRIATE PAYMENT TO: ACOUSTICAL SOCIETY OF AMERICA, 1305 WALT WHITMAN ROAD, SUITE 300, MELVILLE, NY 11747-4300; FAX: 631-923-2875

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Regional Chapters and Student Chapters

Anyone interested in becoming a member of a regional chapter or in learning if a meeting of the chapter will be held while he/she is in the local area of the chapter, either permanently or on travel, is welcome to contact the appropriate chapter representative. Contact information is listed below for each chapter representative.

Anyone interested in organizing a regional chapter in an area not covered by any of the chapters below is invited to contact the Cochairs of the Committee on Regional Chapters for information and assistance: Evelyn Hoglund, Ohio State University, hoglund1@osu.edu and Kenneth W. Good, Jr., Armstrong World Industries, Inc., kwgoodjr@armstrong.com

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www.asala.org

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asdougl@umich.edu

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to papers presented at

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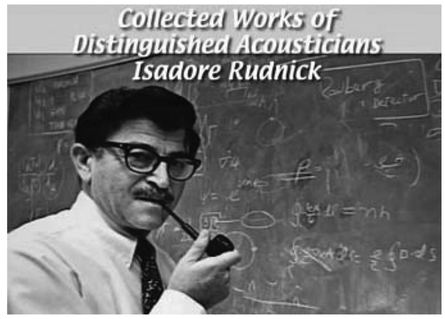
Collected Works of Distinguished Acousticians

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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