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Acoustical Society of America

Wyndham Grand Pittsburgh Downtown Hotel
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18–22 May 2015

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TECHNICAL PROGRAM SUMMARY
18–22 May 2015
*Indicates Special Session

Monday morning, 18 May

*1aAA	Spherical Array Processing for Architectural Acoustics
1aAB	Bioacoustics and Behavior
*1aED	Hands-On Acoustics Demonstrations for Middle- and High-School Students
1aPA	General Topics in Physical Acoustics I
*1aPPa	Kinematic Hearing: Auditory Perception of Moving Sounds by Moving Listeners
1aPPb	General Topics in Psychological Acoustics (Poster Session)
*1aSC	Listening Effort I
1aUW	Acoustic Communications and Scattering

Monday afternoon, 18 May

*1pAAa	Uncertainty in Laboratory Building Acoustics Measurements
*1pAAb	Session in Honor of Laymon Miller
1pABa	Physiology of Animal Hearing
1pABb	Advances in Processing Bioacoustics Data
*1pPA	Acoustofluidics: Interaction of Acoustics and Fluid Dynamic Phenomena
1pPP	Psychoacoustics (Poster Session)
*1pSAa	Blind Source Localization and Separation for Structural Acoustic Sources
1pSAb	General Topics In Structural Acoustics and Vibration I
*1pSC	Listening Effort II
*1pSP	Telecom and Audio Signal Processing
1pUW	Environmental Characterization, Localization and Vectors

Monday evening, 18 May

*1eED	Listen Up and Get Involved
*1eID	Tutorial Lecture on Man-Made Noise and Aquatic Life: Data, Data Gaps, and Speculation

Tuesday morning, 19 May

*2aAAa	Sixty-Fifth Anniversary of Noise and Health: Session in Honor of Karl Kryter I
*2aAAb	Student Design Competition
*2aAB	Auditory Scene Analysis in Natural Environments
*2aBA	Ultrasound Contrast Agents: Nonlinear Bubble Dynamics
*2aNSa	Noise, Soundscape, and Health
*2aNSb	Active Control of Sound and Vibration
*2aPA	Wind Noise Mitigation in Atmospheric Acoustics
*2aPP	Celebration of the Modern Cochlear Implant and the First Substantial Restoration of a Human Sense Using a Medical Intervention I
*2aSA	Acoustic Metamaterials I
2aSC	General Topics in Speech Production (Poster Session)
2aSP	Characteristic Spectral and Temporal Patterns in Speech Signals
*2aUW	Historical Perspectives on the Origins of Underwater Acoustics I

Tuesday afternoon, 19 May

*2pAA	Sixty-Fifth Anniversary of Noise and Health: Session in Honor of Karl Kryter II
*2pAO	Acoustics of Fine Grained Sediments: Theory and Measurements
*2pBA	Ultrasonic Characterization of Bone
2pED	Topics in Acoustics Education
*2pNSa	Application of Psychoacoustics in Noise
*2pNSb	Mobile Technology Solutions for Hearing Loss Prevention
2pPA	General Topics in Physical Acoustics II
*2pPP	Celebration of the Modern Cochlear Implant and the First Substantial Restoration of a Human Sense Using a Medical Intervention II
*2pSAa	Acoustic Metamaterials II
*2pSAb	Time-Domain Methods
2pSC	Speech Methods, Models and Technology (Poster Session)
*2pUW	Historical Perspectives on the Origins of Underwater Acoustics II

Wednesday morning, 20 May

*3aAAa	Public Policy Implementation of School Acoustics
*3aAAb	Session in Honor of Dick Campbell
*3aBA	Acoustic Radiation Force in Biomedical Applications I
*3aED	Preparing Graduate Students for Careers in Acoustics
*3aMUa	Acoustics of Percussion Instruments
*3aMUb	Acoustics of Percussion Instruments II: Concert
*3aNS	Noise with Tonal Components in the Built Environment
3aPP	Spatial Hearing and Localization
3aSA	General Topics in Structural Acoustics and Vibration II
*3aSC	Celebration of Kenneth N. Stevens' Contributions to Speech Communication: Past, Present, Future I
*3aSPa	Smartphone Acoustic Signal Processing Student Competition (Poster Session)
*3aSPb	Cognitive Signal Processing
*3aUW	Historical Perspectives on the Origins of Underwater Acoustics III

Wednesday afternoon, 20 May

*3pAA	Uses, Measurements, and Advances in the Use of Diffusion and Scattering Devices
3pAB	Biosonar
*3pAO	Acoustical Oceanography Prize Lecture
*3pBA	Biomedical Acoustics Best Student Paper Competition (Poster Session)
*3pID	Hot Topics in Acoustics
3pMU	General Topics in Musical Acoustics I
3pNS	Environmental Noise and Noise Control Elements
*3pPP	Auditory Neuroscience Prize Lecture
3pSA	General Topics in Structural Acoustics and Vibration III
*3pSC	Celebration of Kenneth N. Stevens' Contributions to Speech Communication: Past, Present, Future II

Thursday morning, 21 May

4aAAa	Architectural Acoustics Potpourri
*4aAAb	Up and Coming Architectural Acousticians: Past Student Paper Award Winners Report
4aAO	General Topics in Acoustical Oceanography
*4aBA	Acoustic Radiation Force in Biomedical Applications II
*4aEA	Funding Opportunities with the National Science Foundation
*4aED	Expanding Acoustics Outreach with Social Media
*4aNS	Wind Turbine Noise I
*4aPA	Infrasound I
*4aPP	Influence of Visual Cues on Auditory Perception
*4aSA	Noise Identification and Control in the Mining Industry
4aSC	Cross-Language Speech Production and Perception (Poster Session)
4aSP	Beamforming, Optimization, Source Localization and Separation
*4aUW	Three-Dimensional Underwater Acoustics Models and Experiments I

Thursday afternoon, 21 May

4pAA	Classroom and Other Room Acoustics
*4pAB	Audio Playback in Animal Bioacoustics
*4pBAa	Ultrasound Contrast Agents: Molecular Imaging Applications
4pBAb	Therapeutic Ultrasound and Bioeffects
4pEA	General Topics in Engineering Acoustics
4pMU	General Topics in Musical Acoustics II
*4pNS	Wind Turbine Noise II
*4pPA	Infrasound II
4pPP	Physiology, Behavior, and Modeling: From Middle-Ear to Mid-Brain
*4pSA	Demonstrations of Structural Acoustics and Vibration
4pSC	Indexical Factors in Speech Perception and Production (Poster Session)
*4pUW	Three-Dimensional Underwater Acoustics Models and Experiments II

Friday morning, 22 May

5aBA	Medical Ultrasound
*5aMU	Non-Western Musical Instruments and Performance Spaces
*5aNS	Louis C. Sutherland's Lifetime Contributions to the Fields of Noise, Standardization, and Classroom Acoustics
5aSC	Speech Perception and Production in Noise and Related to Disorders of Speech, Language or Hearing (Poster Session)
5aSP	Detection, Classification and Analysis
5aUW	Propagation, Tomography, Scattering and Transducers

ASA Exhibit

at the 2015 Fall Meeting of the Acoustical Society of America

2-6 November

Jacksonville, Florida

The 170th Meeting of the Acoustical Society of America will bring together experts from all fields of acoustics and will provide a forum for the open exchange of scientific information.

The **170th ASA Meeting** will consist of plenary lectures, invited and contributed papers, poster presentations and exhibits. Topics to be covered include:

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

The ASA Meeting will be highlighted by an exhibit featuring displays with instruments, materials, and services for the acoustics and vibration community. The exhibit will be conveniently located near the registration area and meeting rooms and will open with a reception on Monday evening, 2 November, and will close Wednesday, 4 November, at noon.

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

Companies providing products or services related to vibration and acoustics including: computer-based instrumentation, sound level meters, sound intensity systems, signal processing systems, devices for noise control, sound prediction software, acoustical materials, passive and active noise are encouraged to participate in the ASA Exhibits

For information on the **ASA Meeting**:

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	M am	M pm	M eve	Tu am	Tu pm	Tu ev	W am	W pm	Th am	Th pm	Th ev	Fri am
Ballroom 1	1aED 10:00		1eED 5:30	2aSP 8:00	2pNSb 3:30		3aMUa 9:00 3aMUb 11:00					
Ballroom 2	1aPPb 8:00	1pPP 1:00		2aAAb 9:30 2aSC 8:00	2pSC 1:00		3aSPa 8:30	3pBA 1:00	4aSC 8:00	4pSC 1:00		5aSC 8:00
Ballroom 3	1aAA 8:00	1pAAa 1:00	1eID 7:00	2aAAa 8:00	2pAA 1:00	TCAA 7:30	3aAAa 8:30 3aAAb 10:25	3pAA 1:15	4aAAb 9:45	4pAA 1:15	TCSP 8:30	
Ballroom 4	1aUW 8:30	1pUW 1:00		2aUW 8:00	2pUW 1:00	TCPA 7:30	3aUW 9:00	3pID 1:00	4aUW 8:00	4pUW 1:25	TCUW 8:00	5aUW 8:00
Brigade						TCEA 4:30 TCAB 7:30	3aSPb 9:00		4aSP 9:00	4pMU 1:30	TCMU 8:00	
Commonwealth 1				2aNSa 8:30	2pBA 1:00	TCSA 7:30			4aEA 10:00			
Commonwealth 2		1pAAb 2:00			2pED 1:30 2pSAb 3:30	TCPP 7:30	3aSC 8:00	3pSC 2:15	4aAAa 8:00	4pEA 1:00	TCSC 8:00	
Kings 1	1aPA 8:30	1pPA 1:00		2aPA 8:00	2pPA 1:00		3aED 8:30	3pAO 1:00	4aPA 8:30	4pPA 2:00		5aSP 8:30
Kings 2				2aBA 8:00			3aBA 8:00	3pMU 1:00	4aED 8:00	4pBAa 2:00 4pBAb 3:15	TCBA 8:00	5aBA 8:00
Kings 3	1aPPa 8:00	1pSAa 1:00 1pSAb 3:00		2aSA 9:00	2pSAa 1:30		3aSA 8:00	3pSA 1:00	4aSA 8:00	4pSA 2:00		
Kings 4	1aSC 9:00	1pSC 1:30		2aPP 8:30	2pPP 1:00		3aPP 8:00	3pPP 1:15	4aPP 8:00	4pPP 1:30		5aMU 8:30
Kings 5		1pSP 1:00		2aNSb 9:00	2pNSa 1:30		3aNS 8:00	3pNS 2:00	4aNS 8:30	4pNS 2:00	TCNS 8:00	5aNS 8:30
Rivers	1aAB 8:30	1pABa 1:00 1pABb 2:45		2aAB 8:00	2pAO 1:00	TCAO 7:30		3pAB 1:30	4aAO 8:30	4pAB 2:00		
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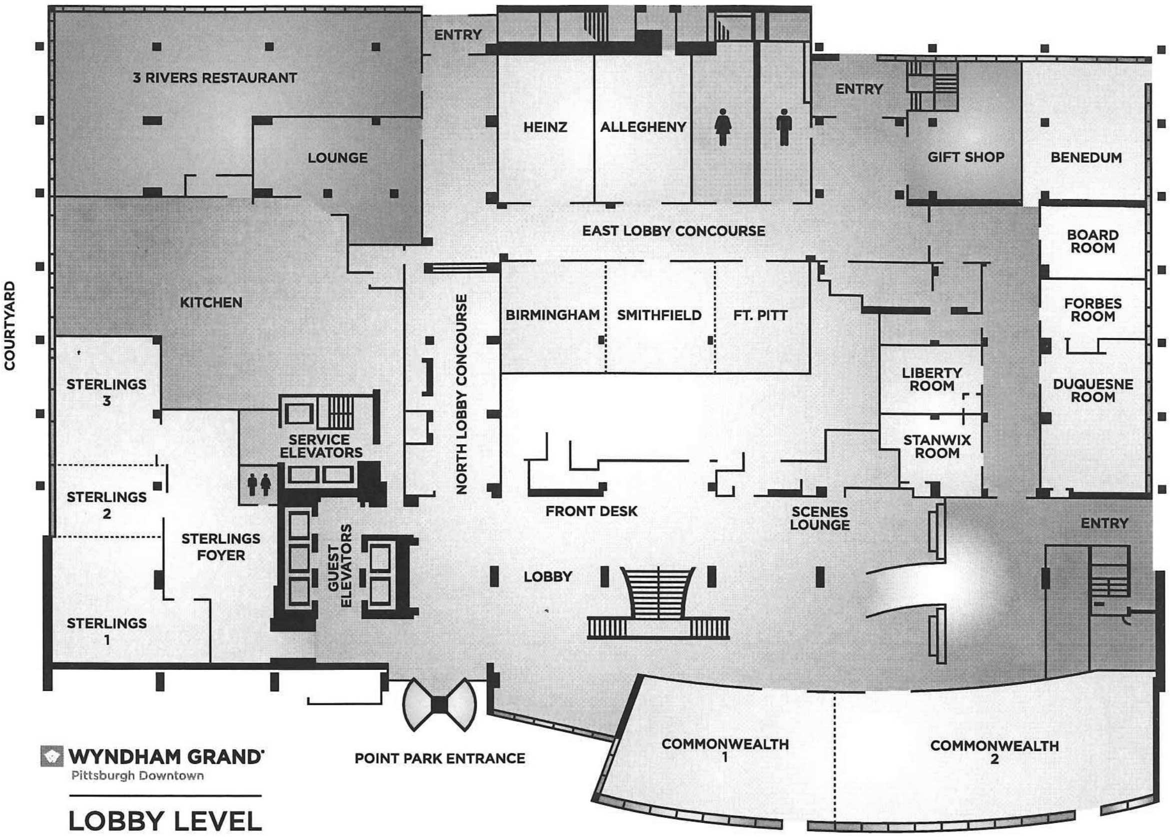
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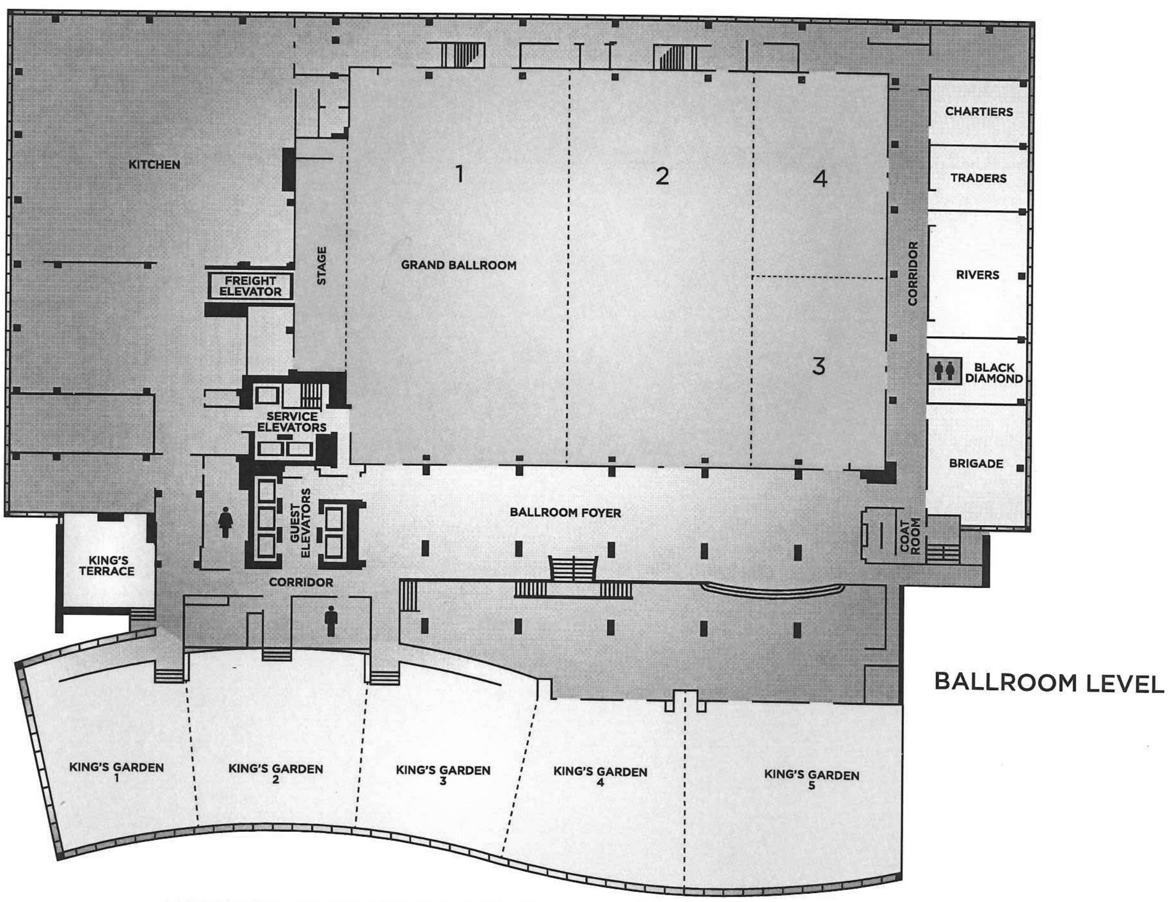
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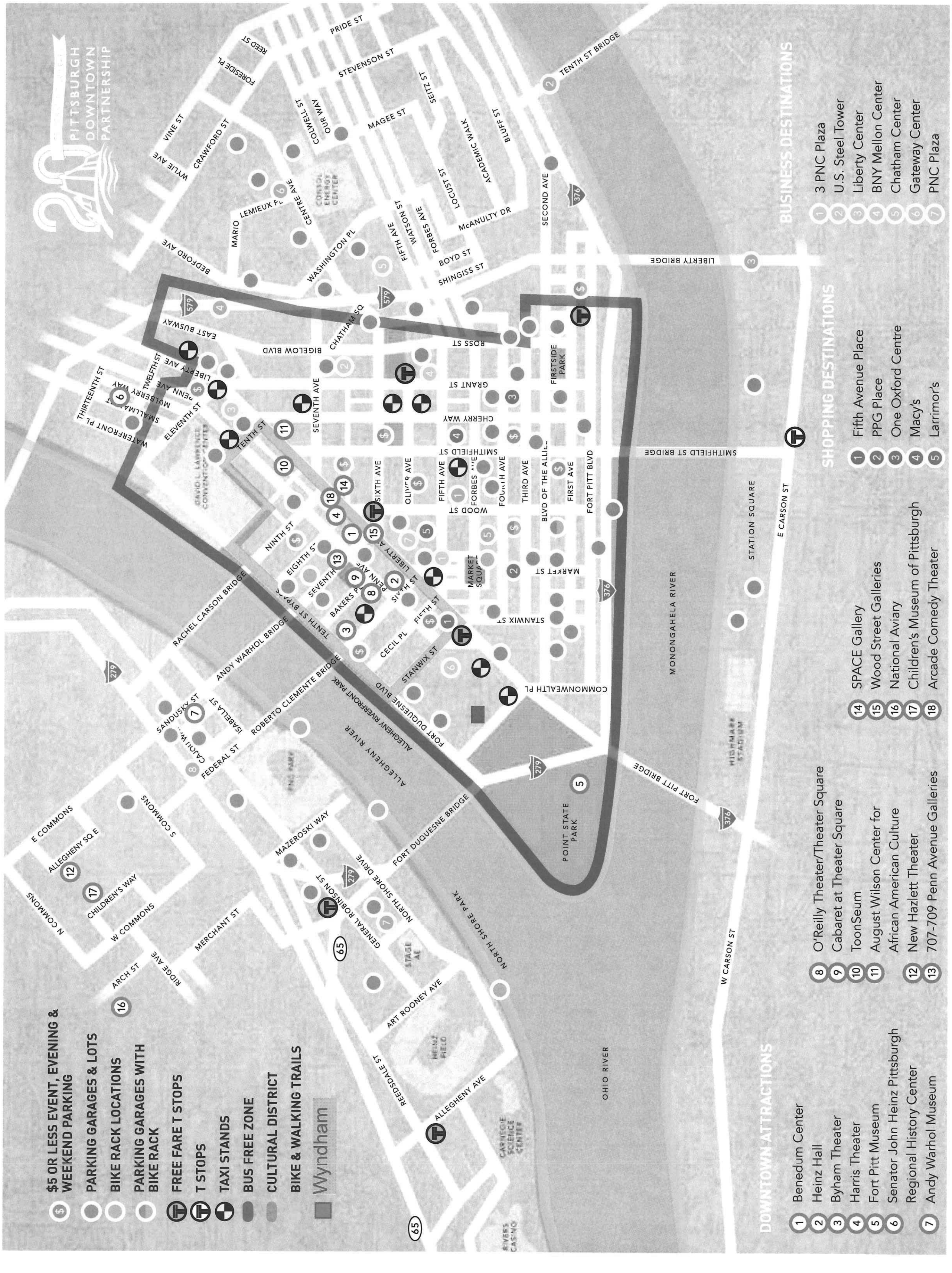

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POINT STATE PARK
STATION SQUARE
MARKET SQUARE
FIRSTSIDE PARK

TECHNICAL PROGRAM CALENDAR
169th Meeting
18–22 May 2015

MONDAY MORNING

8:00 1aAA **Architectural Acoustics, Noise, and Signal Processing in Acoustics:** Spherical Array Processing for Architectural Acoustics. Ballroom 3

8:30 1aAB **Animal Bioacoustics:** Bioacoustics and Behavior. Rivers

10:00 1aED **Education in Acoustics:** Hands-On Acoustics Demonstrations for Middle- and High-School Students. Ballroom 1

8:30 1aPA **Physical Acoustics:** General Topics in Physical Acoustics I. Kings 1

8:00 1aPPa **Psychological and Physiological Acoustics and Animal Bioacoustics:** Kinematic Hearing: Auditory Perception of Moving Sounds by Moving Listeners. Kings 3

8:00 1aPPb **Psychological and Physiological Acoustics:** General Topics in Psychological Acoustics (Poster Session). Ballroom 2

9:00 1aSC **Speech Communication and Psychological and Physiological Acoustics:** Listening Effort I. Kings 4

8:30 1aUW **Underwater Acoustics:** Acoustic Communications and Scattering. Ballroom 4

MONDAY AFTERNOON

1:00 1pAAa **Architectural Acoustics and Noise:** Uncertainty in Laboratory Building Acoustics Measurements. Ballroom 3

2:00 1pAAb **Architectural Acoustics, Noise, and Structural Acoustics and Vibration:** Session in Honor of Laymon Miller. Commonwealth 2

1:00 1pABa **Animal Bioacoustics:** Physiology of Animal Hearing. Rivers

2:45 1pABb **Animal Bioacoustics:** Advances in Processing Bioacoustic Data. Rivers

1:00 1pPA **Physical Acoustics:** Acoustofluidics: Interaction of Acoustics and Fluid Dynamic Phenomena. Kings 1

1:00 1pPP **Psychological and Physiological Acoustics:** Psychoacoustics (Poster Session). Ballroom 2

1:00 1pSAa **Structural Acoustics and Vibration, Noise, and Signal Processing in Acoustics:** Blind Source Localization and Separation for Structural Acoustic Sources. Kings 3

3:00 1pSAb **Structural Acoustics and Vibration:** General Topics in Structural Acoustics and Vibration I. Kings 3

1:30 1pSC **Speech Communication and Psychological and Physiological Acoustics:** Listening Effort II. Kings 4

1:00 1pSP **Signal Processing in Acoustics, Architectural Acoustics, and Noise:** Telecom and Audio Signal Processing. Kings 5

1:00 1pUW **Underwater Acoustics:** Environmental Characterization, Localization and Vectors. Ballroom 4

MONDAY EVENING

5:30 1eED **Education in Acoustics and Women in Acoustics:** Listen Up and Get Involved. Ballroom 1

7:00 1eID **Interdisciplinary:** Tutorial Lecture on Man-Made Noise and Aquatic Life: Data, Data Gaps, and Speculation. Ballroom 3

TUESDAY MORNING

8:00 2aAAa **Architectural Acoustics, Noise, Speech Communication, Psychological and Physiological Acoustics, and Signal Processing:** Sixty-Fifth Anniversary of Noise and Health: Session in Honor of Karl Kryter I. Ballroom 3

9:30 2aAAb **Architectural Acoustics and National Council of Acoustical Consultants:** Student Design Competition. Ballroom 2

8:00 2aAB **Animal Bioacoustics and Psychological and Physiological Acoustics:** Auditory Scene Analysis in Natural Environments. Rivers

8:00 2aBA **Biomedical Acoustics:** Ultrasound Contrast Agents: Nonlinear Bubble Dynamics. Kings 2

8:30 2aNSa **Noise, Psychological and Physiological Acoustics, and ASA Committee on Standards:** Noise, Soundscape, and Health. Commonwealth 1

9:00 2aNSb **Noise and Structural Acoustics and Vibration:** Active Control of Sound and Vibration. Kings 5

8:00 2aPA **Physical Acoustics:** Wind Noise Mitigation in Atmospheric Acoustics. Kings 1

8:30 2aPP **Psychological and Physiological Acoustics, Biomedical Acoustics, Speech Communication, and Signal Processing in Acoustics:** Celebration of the Modern Cochlear Implant and the First Substantial Restoration of a Human Sense Using a Medical Intervention I. Kings 4

9:00 2aSA **Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics:** Acoustic Metamaterials I. Kings 3

8:00 2aSC **Speech Communication:** General Topics in Speech Production (Poster Session). Ballroom 2

8:00 2aSP **Signal Processing in Acoustics, Speech Communication, and Psychological and Physiological Acoustics:** Characteristic Spectral and Temporal Patterns in Speech Signals. Ballroom 1

8:00 2aUW **Underwater Acoustics:** Historical Perspectives on the Origins of Underwater Acoustics I. Ballroom 4

TUESDAY AFTERNOON

1:00 2pAA **Architectural Acoustics, Noise, Speech Communication, Psychological and Physiological Acoustics, and Signal Processing in Acoustics:** Sixty-Fifth Anniversary of Noise and Health: Session in Honor of Karl Kryter II. Ballroom 3

1:00 2pAO **Acoustical Oceanography and Underwater Acoustics:** Acoustics of Fine Grained Sediments: Theory and Measurements. Rivers

1:00 2pBA **Biomedical Acoustics:** Ultrasonic Characterization of Bone. Commonwealth 1

1:30 2pED **Education in Acoustics:** Topics in Acoustics Education. Commonwealth 2

1:30 2pNSa **Noise and Psychological and Physiological Acoustics:** Application of Psychoacoustics in Noise. Kings 5

3:30 2pNSb **Noise and Psychological and Physiological Acoustics:** Mobile Technology Solutions for Hearing Loss Prevention. Ballroom 1

1:00 2pPA **Physical Acoustics:** General Topics in Physical Acoustics II. Kings 1

1:00 2pPP **Psychological and Physiological Acoustics, Biomedical Acoustics, Speech Communication, and Signal Processing in Acoustics:** Celebration of the Modern Cochlear Implant and the First Substantial Restoration of a Human Sense Using a Medical Intervention II. Kings 4

1:30 2pSAa **Structural Acoustics and Vibration, Physical Acoustics, and Engineering Acoustics:** Acoustic Metamaterials II. Kings 3

3:30 2pSAb **Structural Acoustics and Vibration and Signal Processing in Acoustics:** Time-Domain Methods. Commonwealth 2

1:00 2pSC **Speech Communication:** Speech Methods, Models and Technology (Poster Session). Ballroom 2

1:00 2pUW **Underwater Acoustics:** Historical Perspectives on the Origins of Underwater Acoustics II. Ballroom 4

WEDNESDAY MORNING

8:30 3aAAa **Architectural Acoustics, ASA Committee on Standards, and Noise:** Public Policy Implementation of School Acoustics. Ballroom 3

10:25 3aAAb **Architectural Acoustics and Noise:** Session in Honor of Dick Campbell. Ballroom 3

8:00 3aBA **Biomedical Acoustics:** Acoustic Radiation Force in Biomedical Applications I. Kings 2

8:30 3aED **Education in Acoustics and Physical Acoustics:** Preparing Graduate Students for Careers in Acoustics. Kings 1

9:00 3aMUa **Musical Acoustics:** Acoustics of Percussion Instruments. Ballroom 1

11:00 3aMUb **Musical Acoustics:** Acoustics of Percussion Instruments II: Concert. Ballroom 1

8:00 3aNS **Noise, Architectural Acoustics, and ASA Committee on Standards:** Noise with Tonal Components in the Built Environment. Kings 5

8:00 3aPP **Psychological and Physiological Acoustics:** Spatial Hearing and Localization. Kings 4

8:00 3aSA **Structural Acoustics and Vibration:** General Topics in Structural Acoustics and Vibration II. Kings 3

8:00 3aSC **Speech Communication:** Celebration of Kenneth N. Stevens' Contributions to Speech Communication: Past, Present, Future I. Commonwealth 2

8:30 3aSPa **Signal Processing in Acoustics:** Smartphone Acoustic Signal Processing Student Competition (Poster Session). Ballroom 2

9:00 3aSPb **Signal Processing in Acoustics and Psychological and Physiological Acoustics:** Cognitive Signal Processing. Brigade

9:00 3aUW **Underwater Acoustics:** Historical Perspectives on the Origins of Underwater Acoustics III. Ballroom 4

WEDNESDAY AFTERNOON

1:15 3pAA **Architectural Acoustics:** Uses, Measurements, and Advances in the Use of Diffusion and Scattering Devices. Ballroom 3

1:30 3pAB **Animal Bioacoustics:** Biosonar. Rivers

1:00 3pAO **Acoustical Oceanography:** Acoustical Oceanography Prize Lecture. Kings 1

1:00 3pBA **Biomedical Acoustics:** Biomedical Acoustics Best Student Paper Competition (Poster Session). Ballroom 2

1:00 3pID **Interdisciplinary:** Hot Topics in Acoustics. Ballroom 4

1:00 3pMU **Musical Acoustics:** General Topics in Musical Acoustics I. Kings 2

2:00 3pNS **Noise:** Environmental Noise and Noise Control Elements. Kings 5

1:15 3pPP **Psychological and Physiological Acoustics:** Auditory Neuroscience Prize Lecture. Kings 4

1:00 3pSA **Structural Acoustics and Vibration:** General Topics in Structural Acoustics and Vibration III. Kings 3

2:15 3pSC **Speech Communication:** Celebration of Kenneth N. Stevens' Contributions to Speech Communication: Past, Present, Future II. Commonwealth 2

THURSDAY MORNING

8:00 4aAAa **Architectural Acoustics:** Architectural Acoustics Potpourri. Commonwealth 2

9:45 4aAAb **Architectural Acoustics and Education in Acoustics:** Up and Coming Architectural Acousticians: Past Student Paper Award Winners Report. Ballroom 3

8:30 4aAO **Acoustical Oceanography:** General Topics in Acoustical Oceanography. Rivers

9:30 4aBA **Biomedical Acoustics:** Acoustic Radiation Force in Biomedical Applications II. Sterlings 1/2

10:00 4aEA **Engineering Acoustics and Physical Acoustics:** Funding Opportunities with the National Science Foundation. Commonwealth 1

8:00 4aED **Education in Acoustics Public Relations, and Student Council:** Expanding Acoustics Outreach with Social Media. Kings 2

8:30 4aNS **Noise, ASA Committee on Standards, Psychological and Physiological Acoustics, Physical Acoustics, and Structural Acoustics and Vibration:** Wind Turbine Noise I. Kings 5

8:30 4aPA **Physical Acoustics:** Infrasound I. Kings 1

8:00 4aPP **Psychological and Physiological Acoustics and Speech Communication:** Influence of Visual Cues on Auditory Perception. Kings 4

8:00 4aSA **Structural Acoustics and Vibration and Noise:** Noise Identification and Control in the Mining Industry. Kings 3

8:00 4aSC **Speech Communication:** Cross-Language Speech Production and Perception (Poster Session). Ballroom 2

9:00 4aSP **Signal Processing in Acoustics:** Beamforming, Optimization, Source Localization and Separation. Brigade

8:00 4aUW **Underwater Acoustics:** Three-Dimensional Underwater Acoustics Models and Experiments I. Ballroom 4

THURSDAY AFTERNOON

1:15 4pAA **Architectural Acoustics:** Classroom and Other Room Acoustics. Ballroom 3

2:00 4pAB **Animal Bioacoustics:** Audio Playback in Animal Bioacoustics. Rivers

1:00 4pBAa **Biomedical Acoustics:** Ultrasound Contrast Agents: Molecular Imaging Applications. Kings 2

3:15 4pBAb **Biomedical Acoustics:** Therapeutic Ultrasound and Bioeffects. Kings 2

1:00 4pEA **Engineering Acoustics:** General Topics in Engineering Acoustics. Commonwealth 2

1:30 4pMU **Musical Acoustics:** General Topics in Musical Acoustics II. Brigade

2:00 4pNS **Noise, ASA Committee on Standards, Psychological and Physiological Acoustics, Physical Acoustics, and Structural Acoustics and Vibration:** Wind Turbine Noise II. Kings 5

2:00 4pPA **Physical Acoustics:** Infrasound II. Kings 1

1:30 4pPP **Psychological and Physiological Acoustics:** Physiology, Behavior, and Modeling: From Middle-Ear to Mid-Brain. Kings 4

2:00 4pSA **Structural Acoustics and Vibration, Education in Acoustics, and Physical Acoustics:** Demonstrations of Structural Acoustics and Vibration. Kings 3

1:00 4pSC **Speech Communication:** Indexical Factors in Speech Perception and Production (Poster Session). Ballroom 2

1:25 4pUW **Underwater Acoustics:** Three-Dimensional Underwater Acoustics Models and Experiments II. Ballroom 4

FRIDAY MORNING

8:00 5aBA **Biomedical Acoustics:** Medical Ultrasound. Kings 2

8:30 5aMU **Musical Acoustics:** Non-Western Musical Instruments and Performance Spaces. Kings 4

8:30 5aNS **Noise and ASA Committee on Standards:** Louis C. Sutherland's Lifetime Contributions to the Fields of Noise, Standardization, and Classroom Acoustics. Kings 5

8:00 5aSC **Speech Communication:** Speech Perception and Production in Noise and Related to Disorders of Speech, Language or Hearing (Poster Session). Ballroom 2

8:30 5aSP **Signal Processing in Acoustics:** Detection, Classification, and Analysis. Kings 1

8:00 5aUW **Underwater Acoustics:** Propagation, Tomography, Scattering, and Transducers. Ballroom 4

SCHEDULE OF COMMITTEE MEETINGS AND OTHER EVENTS

ASA COUNCIL AND ADMINISTRATIVE COMMITTEES

Mon, 18 May, 7:30 a.m.	Executive Council	Benedum
Mon, 185 May, 3:30 p.m.	Technical Council	Benedum
Tue, 19 May, 7:00 a.m.	ASA Press Editorial Board	Heinz
Tue, 19 May, 7:00 a.m.	POMA Editorial Board	Allegheny
Tue, 19 May, 7:30 a.m.	Panel on Public Policy	Liberty
Tue, 19 May, 7:30 a.m.	Translation of Chinese Journals	Stanwix
Tue, 19 May, 11:45 a.m.	Editorial Board	Sterling 2/3
Tue, 19 May, 12:00 noon	Activity Kit	Liberty
Tue, 19 May, 12:00 noon	Student Council	Allegheny
Tue, 19 May, 12:30 p.m.	Prizes & Special Fellowships	Stanwix
Tue, 19 May, 1:30 p.m.	Meetings	Benedum
Tue, 19 May, 4:00 p.m.	Books+	Liberty
Tue, 19 May, 4:00 p.m.	Education in Acoustics	Ft. Pitt
Tue, 19 May, 4:30 p.m.	Newman Fund Advisory	Board Room
Tue, 19 May, 5:00 p.m.	Women in Acoustics	Benedum
Wed, 20 May, 6:45 a.m.	International Research & Education	Ft. Pitt
Wed, 20 May, 7:00 a.m.	College of Fellows	Forbes
Wed, 20 May, 7:00 a.m.	Publication Policy	Heinz
Wed, 20 May, 7:00 a.m.	Regional Chapters	Benedum
Wed, 20 May, 10:00 a.m.	Publishing Services	Forbes
Wed, 20 May, 11:00 a.m.	Medals and Awards	Benedum
Wed, 20 May, 11:30 a.m.	Audit	Liberty
Wed, 20 May, 11:30 a.m.	Public Relations	Heinz
Wed, 20 May, 12:00 noon	Membership	Ft. Pitt
Wed, 20 May, 1:30 p.m.	AS Foundation Board	Duquesne
Wed, 20 May, 5:00 p.m.	Acoustics Today Advisory	Duquesne
Thu, 21 May, 7:00 a.m.	Archives & History	Duquesne
Thu, 21 May, 7:30 a.m.	Tutorials	Stanwix
Thu, 21 May, 7:30 a.m.	Investment	Liberty
Thu, 21 May, 2:00 p.m.	Publishing	Board Room
Thu, 21 May, 4:30 p.m.	External Affairs	Duquesne
Thu, 21 May, 4:30 p.m.	Internal Affairs	Benedum
Fri, 22 May, 7:00 a.m.	Technical Council	Benedum
Fri, 22 May, 11:00 a.m.	Executive Council	Benedum

TECHNICAL COMMITTEE OPEN MEETINGS

Tue, 19 May, 4:30 p.m.	Engineering Acoustics	Brigade
Tue, 19 May, 7:30 p.m.	Acoustical Oceanography	Rivers
Tue, 19 May, 7:30 p.m.	Animal Bioacoustics	Brigade
Tue, 19 May, 7:30 p.m.	Architectural Acoustics	Ballroom 3
Tue, 19 May, 7:30 p.m.	Physical Acoustics	Ballroom 4
Tue, 19 May, 7:30 p.m.	Psychological and Physiological Acoustics	Commonwealth 2
Tue, 19 May, 7:30 p.m.	Structural Acoustics and Vibration	Commonwealth 1
Thu, 21 May, 8:00 p.m.	Biomedical Acoustics	Kings 2
Thu, 12 May, 8:00 p.m.	Musical Acoustics	Brigade
Thu, 21 May, 8:00 p.m.	Noise	Kings 5
Thu, 21 May, 8:00 p.m.	Speech Communication	Commonwealth 2
Thu, 21 May, 8:00 p.m.	Underwater Acoustics	Ballroom 4
Thu, 21 May, 8:30 p.m.	Signal Processing in Acoustics	Ballroom 3

STANDARDS COMMITTEES AND WORKING GROUPS

Mon, 18 May, 5:00 p.m.	S2, Mechanical Vibration and Shock	Duquesne
Mon, 18 May, 7:00 p.m.	ASACOS Steering	Duquesne
Tue, 19 May, 7:30 a.m.	ASACOS	Duquesne
Tue, 19 May, 9:15 a.m.	Standards Plenary	Duquesne
Tue, 19 May, 11:00 a.m.	S1, Acoustics	Duquesne
Tue, 19 May, 1:45 p.m.	S3/SC1 Animal Bioacoustics	Duquesne
Tue, 19 May, 3:00 p.m.	S3, Bioacoustics	Duquesne
Tue, 19 May, 4:30 p.m.	S12, Noise	Duquesne
Wed, 20 May, 7:30 a.m.	S12/WG18-Room Criteria	Liberty
Wed, 20 May, 8:00 a.m.	S3/WG35-Pure-Tone Audiometry	Duquesne
Wed, 20 May, 9:00 a.m.	S12/WG56-Soundscapes in Parks	Board Room
Wed, 20 May, 5:00 p.m.	S12/WG44-Speech Privacy	Ft. Pitt

MEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Mon-Thu, 18-21 May 7:30 a.m.-5:00 p.m. Fri, 22 May 7:30 a.m.-12:00 noon	Registration	Ballroom Foyer
Mon-Thu, 18-21 May, 7:00 a.m.-5:00 p.m. Fri, 22 May, 7:00 a.m.-12:00 noon	E-mail Room	Chartiers
Mon-Thu, 3-6 June 8:00 a.m.-5:00 p.m. Fri, 9 May 8:00 a.m.-12:00 noon	Internet Café	Birmingham
Mon-Thu, 18-21 May 7:00 a.m.-5:00 p.m. Fri, 22 May 7:00 a.m.-12:00 noon	A/V Preview	Chartiers
Mon-Thu, 18-21 May 8:00 a.m. to 10:00 a.m.	Accompanying Persons	Sterling 3
Sun, 17 May 1:00 p.m.-5:00 p.m. Mon, 18 May 8:30 a.m.-12:30 p.m.	Short Course	Smithfield
Mon-Fri, 18-22 May 9:45 a.m.-10:30 a.m.	A.M. Coffee Break	Ballroom Foyer
Tue-Thu, 19-21 May 12:00 p.m.-1:00 p.m.	Resume Help Desk	Ballroom Foyer
Mon, 18 May 5:00 p.m.-5:30 p.m.	New Student Orientation	Kings 5
Mon, 18 May 5:30 p.m.-6:45 p.m.	Student Meet and Greet	Sterling 2/3
Mon, 18 May 5:30 p.m.-7:30 p.m.	Technical Tour-Benedum Theater	Wyndham Lobby 5:15 p.m.
Tue, 19 May, 6:00 p.m.-7:30 p.m.	Social Hour	Kings 1-5
Wed, 20 May 9:00 a.m.-11:30 a.m.	Technical Tour: Aviary	Wyndham Lobby 8:30 a.m.
Wed, 20 May, 11:30 a.m.-1:30 p.m.	Women in Acoustics Luncheon	Sterling 1/2/3
Wed, 20 May, 3:30 p.m.-5:00 p.m.	Plenary Session/Awards Ceremony	Ballroom 1
Wed, 20 May, 6:00 p.m.-7:30 p.m.	Student Reception	Commonwealth 2
Wed, 20 May, 6:00 p.m.	Penn State Dinner	Sterling 1/2/3
Wed, 20 May, 8:00 p.m.-12:00 midnight	ASA Jam	Ballroom 1
Thu, 21 May, 12:00 noon-2:00 p.m.	Society Luncheon and Lecture	Ballroom 1
Thu, 7 May, 6:30 p.m.-8:00 p.m.	Social Hour	Gateway Clipper Fleet Dock 6:00 p.m.

169th Meeting of the Acoustical Society of America

The 169th meeting of the Acoustical Society of America will be held Monday through Friday, 18–22 May 2015 at the Wyndham Grand Pittsburgh Downtown Hotel, Pittsburgh, Pennsylvania, USA.

SECTION HEADINGS

1. HOTEL INFORMATION

The Wyndham Grand Pittsburgh Downtown Hotel is the headquarters hotel where all meeting events will be held.

The cut-off date for reserving rooms at special rates has passed. Please contact the Wyndham Grand Pittsburgh Downtown Hotel for reservation information: 600 Commonwealth Place, Pittsburgh, PA 15222, T: 412-391-4600; F: 412-467-3474; W: wyndhamgrandpittsburgh.com

2. TRANSPORTATION AND TRAVEL DIRECTIONS

Pittsburgh is served by many major airlines through the Pittsburgh International Airport which is located 19 miles from downtown Pittsburgh. For flight information visit www.pitairport.com.

Transportation from the Pittsburgh International Airport to Wyndham Grand Pittsburgh Downtown Hotel and Downtown Pittsburgh area hotels:

INFORMATION: Desks are located in both the Landside and Airside Terminals. Use the white courtesy phones located throughout the airport to reach customer service representatives by dialing 6.

AIRPORT SHUTTLE SHARED-RIDE SERVICE: One way to/from Pittsburgh International Airport to Wyndham Grand Pittsburgh Downtown is USD $24.00 and a pre-purchased round trip fare is USD $46.00. SuperShuttle Pittsburgh, (800) 258-3826 www.supershuttle.com.

TAXICABS AND LIMOUSINES: Taxis and limousines are available at the Landside terminal outside of baggage claim. Downtown Pittsburgh is approximately 25 minutes from the airport, with fares averaging $40.00-45.00 USD one way. For more information visit; www.pitairport.com or phone (412) 472-5538.

MAJOR CAR RENTAL COMPANIES: Rental counters are on the baggage claim level of the Landside Terminal.

PARKING AT THE WYNDHAM GRAND PITTSBURGH DOWNTOWN HOTEL: Valet parking at the hotel is first come, first served. The overnight charge is $30.00 per night and availability is limited. Current day parking charges are $22.00 per day. To view parking garages in the area and their rates visit, apps.pittsburghpa.gov.

Pittsburgh can also be reached by rail and by bus:

RAIL SERVICE: Amtrak's Pittsburgh (PGH) Station is located at 1100 Liberty Avenue, Pittsburgh, less than 1 mile from the Wyndham Grand Pittsburgh Downtown Hotel. To view routes please visit www.amtrak.com.

BUS SERVICE: Greyhound bus has an Intermodal Station less than 1 mile from the Wyndham Grand Pittsburgh Downtown hotel at 55 11th St., Pittsburgh, PA. Phone (412) 392-6514 or visit www.greyhound.com for rates and reservations.

3. STUDENT TRANSPORTATION SUBSIDIES

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

4. MESSAGES FOR ATTENDEES

A message board will be available in the registration area for attendees to post messages for other attendees.

5. REGISTRATION

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

Registration will open on Monday, 18 May, at 7:30 a.m. in the Grand Ballroom Foyer (see floor plan on page A11).

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

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

One-day registration is available at $295 for members and $360 for nonmembers (one-day means attending the meeting on only one day either to present a paper and/or to attend sessions). A nonmember who pays the $645 nonmember registration fee and simultaneously applies for Associate Membership in the Acoustical Society of America will be given a $50 discount off their dues payment for 2015 dues.

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

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

6. ASSISTIVE LISTENING DEVICES

Any attendee who will require an assistive listening device should advise the Society in advance of the meeting by writing to: Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300; asa@acousticalsociety.org

7. TECHNICAL SESSIONS

The technical program includes 97 sessions with 945 papers scheduled for presentation during the meeting.

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

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

8. TECHNICAL SESSION DESIGNATIONS

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

1-Monday, 18 May
2-Tuesday, 19 May
3-Wednesday, 20 May
4-Thursday, 21 May
5-Friday, 22 May

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

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

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

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

9. HOT TOPICS SESSION

Hot Topics session 3pID will be held on Wednesday, 20 May, at 1:00 p.m. in Ballroom 4. Papers will be presented on current topics in the fields of Noise, Physical Acoustics, and Speech Communication.

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

The 2015 Medwin Prize in Acoustical Oceanography will be awarded to Karim G. Sabra, Georgia Institute of Technology, at the Plenary Session on Wednesday, 20 May.

Karim Sabra will present the Acoustical Oceanography Prize Lecture titled "Monitoring deep ocean temperatures using low-frequency ambient noise" on Wednesday, 20 May, at 1:00 p.m. in Session 3pAO in Kings1.

11. WILLIAM AND CHRISTINE PRIZE IN AUDITORY NEUROSCIENCE AND THE AUDITORY NEUROSCIENCE PRIZE LECTURE

The 2015 William and Christine Prize in Auditory Neuroscience will be awarded to Laurel H. Carney, University of Rochester, at the Plenary Session on Wednesday, 20 May. Laurel Carney will present the Auditory Neuroscience Prize Lecture titled "Relating physiology to perception: The case of the notched-noise masker" on Wednesday, 20 May, at 1:15 p.m. in Session 3pPP in Kings 4.

12. TUTORIAL LECTURE

A tutorial presentation on "Man-Made Noise and Aquatic Life: Data, Data Gaps, and Speculation" will be given by Arthur N. Popper, Professor Emeritus at the University of Maryland (College Park) on Monday, 18 May at 7:00 p.m. in Ballroom 3. This talk will consider how man-made sounds may impact aquatic life – with a focus on fishes (with possible digressions to invertebrates, marine mammals, and turtles). Lecture notes will be available at the meeting in limited supply. To partially defray the cost of the lecture, a registration fee is charged. The fee is USD$25. Students with current ID cards is USD$7.

13. SHORT COURSE

A short course on Time Series Analysis and Adaptive Signal Processing in Acoustics will be given in two parts: Sunday, 17 May, from 1:00 p.m. to 5:00 p.m. and Monday, 18 May, from 8:30 a.m. to 12:30 p.m. in the Smithfield Room.

The processing and analysis of signals are fundamental to most disciplines in the broad field of acoustics. This course will cover the development and analysis of the common algorithms and approaches used for the processing and analysis of time series in acoustics. The theoretical basis of the algorithms as well as their implementation and performance will be covered. The course assumes an advanced undergraduate level of understanding of probability and random processes, linear algebra, and linear dynamic systems.

The course instructor is James Preisig of JPAnalytics LLC. Dr. Preisig is a Senior Member of the IEEE and a Fellow of the ASA. He serves on the Signal Processing in Acoustics and Underwater Acoustics Technical Committees and the Membership Committee of the ASA.

The full registration fee is USD$300 (USD$125 for students) and covers attendance, instructional materials and coffee breaks. The number of attendees is limited to 30.

14. STUDENT DESIGN COMPETITION

The 2015 Student Design Competition will be held on Tuesday, 19 May, in session 2aAAb at 9:30 a.m. in Ballroom 2. This competition is intended to encourage students in the disciplines of architecture, engineering, physics, and other curricula that involve building design and/or acoustics to express their knowledge of architectural acoustics and noise control in the design of a facility in which acoustical considerations are of significant importance. The Student Design Competition is sponsored by the ASA Technical Committee on Architectural Acoustics, with support from the Wenger Foundation, the Robert Bradford Newman Student Award Fund, and the National Council of Acoustical Consultants.

15. RESUME HELP DESK

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

16. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday at 7:30 p.m. and Thursday at 8:00 p.m. at the Wyndham Grand Pittsburgh Downtown Hotel. The meetings on Tuesday and Thursday will be held in the evenings after the socials, except Engineering Acoustics which will meet at 4:30 p.m. on Tuesday. The schedule and rooms for each Committee meeting are given on page A17.

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

17. TECHNICAL TOURS

Monday, 18 May, 5:30 p.m. to 7:30 p.m. Walking tour of Performing Arts Spaces. Tour fee: USD $15 A walking tour of Performing Arts spaces such as the Benedum Center, Heinz Hall, O'Reilly Theatre, with additions to be announced, will be held on Monday, 18 May 2015, 5:30 p.m. to 7:30 p.m. The tour will be led by Christopher M. Evans, House Sound Engineer, Benedum Center. Space is limited to 30 participants. The tour will meet in the Wyndham's first floor lobby at 5:15 p.m. for the 0.5 mi walk to the Benedum Center. Full mobility is required as some of the spaces' rear entrances and exits are not handicapped accessible. Inquire at the registration desk for availability of tour tickets.

Wednesday, 20 May, 9:00 a.m. to 11:30 a.m. (bus boarding at the hotel is 8:30 a.m. and at the aviary at 11:30 a.m.). Tour fees: USD $40 includes transportation. Limited to 30 participants.

The National Aviary is America's only independent indoor nonprofit zoo dedicated to birds. The Aviary's collection features birds representing every continent except Antarctica. Many of these species are showcased in free-flight mixed species exhibits, to allow the birds to demonstrate natural behaviors. Avian expert, Dave Miller, will present his research on flamingo vocalizations and behavioral responses to auditory stimuli and provide an onsite demonstration. Dave will then lead a private tour of the facility, focusing on vocalizations of the Aviary's unparalleled collection of bird species. Participants may bring small recording devices to capture an exceptional array of unique bird calls. Tour participants should gather in the hotel lobby at 8:30 a.m. for bus boarding. Inquire at the registration desk for availability of tour tickets.

Start times are when the bus leaves the hotel, so plan on being there ahead of time.

18. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 20 May, at 3:30 p.m. in Ballroom 1.

The Medwin Prize in Acoustical Oceanography will be presented to Karim S. Sabra and the William and Christine Prize in Auditory Neuroscience will be presented to Laurel H. Carney. Lily Wang, recipient of the ASA Student Council Mentoring Award and Aaron Moberly, recipient of the Research Grant in Speech Research of the American Speech Language and Hearing Foundation will be introduced. The recipients of the American Institute of Physics Prize for Industrial Applications of Physics will be presented to Richard Ruby, John Larson, and Paul Bradley.

The R. Bruce Lindsay Award will be presented to Matthew W. Urban, the Helmholtz-Rayleigh Interdisciplinary Silver Medal will be presented to Henry E. Cox, and the Gold Medal will be presented to Gerhard M. Sessler.

Certificates will be presented to Fellows elected at the Indianapolis meeting of the Society. See page 2344 for a list of fellows.

19. ANSI STANDARDS COMMITTEES

Meetings of ANSI Accredited Standards Committees will be held at the Pittsburgh meeting on the days and times listed in the Schedule of Committee Meetings and Other Events on page A17.

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

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

20. COFFEE BREAKS

Morning coffee breaks will be held each day from 9:45 a.m. to 10:30 a.m. in the Grand Ballroom Foyer.

21. A/V PREVIEW ROOM

The Chartiers Room on the ballroom level 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.

22. PROCEEDINGS OF MEETINGS ON ACOUSTICS (POMA)

The Pittsburgh meeting will have a published proceedings, and submission is optional. This is an open access journal, so that its articles are available in pdf format without charge to anyone in the world for downloading. 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. Further information regarding POMA can be found at the site http://scitation.aip.org/content/asa/journal/poma. Published papers from previous meeting can be seen at the site http://asadl/poma.

23. E-MAIL ACCESS, INTERNET CAFÉ, AND BREAK ROOM

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 Chartiers Room. The Internet Café will be located in the Birmingham Room.

Wifi will be available in all ASA meeting rooms and spaces.

24. SOCIALS

Complimentary buffet socials with cash bar will be held on Tuesday and Thursday evenings.

The social on Tuesday, 19 May, will be held at the Wyndham Hotel from 6:00 p.m. to 7:30 p.m. in Kings 1-5.

The social on Thursday, 21 May, will be held on Gateway Clipper Riverboats cruising the three rivers that converge in Pittsburgh (Allegheny, Monongahela and Ohio). Boats will depart and return to Point State Park which is located directly across from the Wyndham (less than a 5-minute walk). There will be three departures. The first boat, which will have the highest capacity, will board at 5:45 p.m. and sail at 6:00 p.m. It will return at 7:30 p.m. Departure with limited capacities after 6:00 p.m. will be boarded in order of arrival. The open Technical Committee meetings will start at 8:00 p.m. to provide sufficient time for your return to the Wyndham. Please refer to the meeting webpage at acousticalsociety.org/ for updated information.

All meeting attendees are invited to attend the social hours. 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 Technical Committees meetings that begin immediately after the socials.

25. SOCIETY LUNCHEON AND LECTURE

The Society Luncheon and Lecture will be held on Thursday, 21 May, at 12:00 noon in Ballroom 1. The luncheon is open to all attendees and their guests. The speaker is Roger B. Dannenberg, Professor of Computer Science, Art, and Music at Carnegie Mellon University. Purchase your tickets at the Registration Desk before 10:00 a.m. on Wednesday, 21 May. The cost is $30.00 per ticket.

26. STUDENTS MEET MEMBERS FOR LUNCH

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

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

Follow the student twitter @ASAStudents, throughout the meeting.

A New Students Orientation will be held from 5:00 p.m. to 5:30 p.m. on Monday, 18 May, in Sterling 1 for all students to learn about the activities and opportunities available for students at the Pittsburgh meeting. This will be followed by the Student Meet and Greet from 5:30 p.m. to 6:45 p.m. in Sterling 2/3. Refreshments and a cash bar will be available. Students are encouraged to attend the tutorial lecture on Man-Made Noise and Aquatic Life which begins at 7:00 p.m. in Ballroom 3.

The Students' Reception will be held on Wednesday, 19 May, from 6:00 p.m. to 7:30 p.m. in Commonwealth 2. This reception, sponsored by the Acoustical Society of America and supported by the National Council of Acoustical Consultants, will provide an opportunity for students to meet informally with fellow students and other members of the Acoustical Society. All students are encouraged to attend, especially students who are first time attendees or those from smaller universities.

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

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

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

28. WOMEN IN ACOUSTICS LUNCHEON

The Women in Acoustics luncheon will be held at 11:30 a.m. on Wednesday, 20 May, in Sterling 1/2/3 on the lobby level of the Wyndham. Meeting participants who wish to attend must purchase their tickets in advance by 10:00 a.m. on Tuesday, 19 May. The fee is USD$30 for non-students and USD$15 for students.

29. JAM SESSION

You are invited to Ballroom 1 on Wednesday night, 20 May, from 8:00 p.m. to midnight for the JAM SESSION. Bring your axe, horn, sticks, voice, or anything else that makes music. Musicians and non-musicians are all welcome to attend. A full PA system, backline equipment, guitars, bass, keyboard, and drum set will be provided. All attendees will enjoy live music, a cash bar with snacks, and all-around good times. Don't miss out.

30. PENN STATE ACOUSTICS 50TH ANNIVERSARY DINNER

The Penn State Graduate Program in Acoustics is celebrating its 50th Anniversary in 2015, and Penn State is sponsoring a Celebration Reception and Dinner on the evening of Wednesday, 20 May in conjunction with the Pittsburgh ASA meeting. All ASA meeting attendees who self-identify as current or former faculty members, students, alumni, or friends of the Penn State Acoustics Program, and their accompanying persons, are welcome to attend. Although the event will be substantially underwritten by Penn State, there is a nominal charge for the dinner of USD $40. ASA attendees may pay the fee for the dinner at the time of their ASA registration. There will be a separate registration mechanism for current Penn State students, and they should direct inquires directly to the Penn State Acoustics Program offices. For more information, please contact Mrs. Karen J. Thal at (814) 865-6364 or kjt3@psu.edu. Additional details will be available at www.acs.psu.edu. Please come join us for this special 50th Anniversary Celebration. We are ... Penn State!

31. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the Pittsburgh meeting. The registration fee is USD$150.

A hospitality room for accompanying persons will be open at the Wyndham from 8:00 a.m. to 10:00 a.m., Monday through Thursday. Pittsburgh is a beautiful city that has much to see and do.

32. WEATHER

Temperatures average mid 70s during the day dipping into the low 60s in the evening. Pittsburgh does tend to see a bit of light rain this time of year, so don't forget to pack an umbrella!

33. TECHNICAL PROGRAM ORGANIZING COMMITTEE

Jennifer Miksis-Olds, Chair; David P. Knobles, Acoustical Oceanography; Benjamin Taft, Animal Bioacoustics; Damian J. Doria and David T. Bradley, Architectural Acoustics; Siddhartha Sikdar, Biomedical Acoustics; Andrew A. Piacsek, Michelle C. Vigeant, Education in Acoustics; Roger T. Richards, Engineering Acoustics; Andrew C.H. Morrison, Musical Acoustics; Victor W. Sparrow, Noise; Michael R. Haberman, Physical Acoustics; Christopher A. Brown, Psychological and Physiological Acoustics; Ning Xiang, Said Assous, Signal Processing in Acoustics; Alexander L. Francis, Speech Communication; Robert M. Koch, Structural Acoustics and Vibration; Nicholas P. Chotiros, Underwater Acoustics.

34. MEETING ORGANIZING COMMITTEE

Robert Keolian and Matthew Poese, Cochairs; Jennifer Miksis-Olds, Technical Program Chair; Patrick Marcoux, Audio-Visual; Derek Olson, Signs; Gregory Coudriet, Technical Tours; Richard M. Stern, Society Luncheon; Gail Paolino, Meeting Administrator.

35. PHOTOGRAPHING AND RECORDING

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

36. ABSTRACT ERRATA

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

37. GUIDELINES FOR ORAL PRESENTATIONS,

Preparation of Visual Aids

See the enclosed guidelines for computer projection.

- Allow at least one minute of your talk for each slide (e.g., Powerpoint, Keynote, or transparencies). 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 and the text size for labels and axis numbers or letters should be large enough to read.
- 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:
  (1) Statement of research problem
  (2) Research methodology
  (3) Review of results
  (4) 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.

38. 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.

39. GUIDELINES FOR USE OF COMPUTER PROJECTION

- A PC computer with audio playback capability and a 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 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 to prepare their presentation must save their presentations so that the projection works when the presentation is run from the PC in the session room. Also, authors who plan to play audio or video clips during their presentations should insure that their sound (or other) files are also saved on the USB drive and are also uploaded to the PC in the session room. Presenters should also check that the links to the sound (and other) files in the presentation still work after everything has been loaded onto the session room computer.

Using your own computer (only if you really need to!)

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.

General Guidelines

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

SPECIFIC HARDWARE CONFIGURATIONS

Macintosh

- Older Macs require a special adapter to connect the video output port to the standard 15-pin male DIN connector. Make sure you have one with you.
- Hook everything up before powering anything on. (Connect the computer to the RGB input on the projector).

- Turn the projector on and boot up the Macintosh. If this doesn't work immediately, you should make sure that your monitor resolution is set to 1024x768 for an XGA projector or at least 640x480 for an older VGA projector. (1024x768 will most always work.). You should also make sure that your monitor controls are set to mirroring. If it's an older powerbook, it may not have video mirroring, but something called simulscan, which is essentially the same.
- Depending upon the vintage of your Mac, you may have to reboot once it is connected to the computer projector or switcher. Hint: you can reboot while connected to the computer projector in the A/V preview room in advance of your presentation, then put your computer to sleep. Macs thus booted will retain the memory of this connection when awakened from sleep.
- Depending upon the vintage of your system software, you may find that the default video mode is a side-by-side configuration of monitor windows (the test for this will be that you see no menus or cursor on your desktop; the cursor will slide from the projected image onto your laptop's screen as it is moved). Go to Control Panels, Monitors, configuration, and drag the larger window onto the smaller one. This produces a mirror-image of the projected image on your laptop's screen.
- Also depending upon your system software, either the Control Panels will automatically detect the video projector's resolution and frame rate, or you will have to set it manually. If it is not set at a commensurable resolution, the projector may not show an image. Experiment ahead of time with resolution and color depth settings in the A/V preview room (please don't waste valuable time adjusting the Control Panel settings during your allotted session time).

PC

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

Linux

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

40. 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

170th Meeting, Jacksonville, Florida, 2–6 November 2015

171st Meeting, Salt Lake City, Utah, 23–27 May 2016

172nd Meeting, Honolulu, Hawaii, 28 November–2 December 2016

173rd Meeting, Acoustics'17 Boston, Boston, MA, 25–29 June 2017, Joint meeting of the Acoustical Society of America and the European Acoustics Association

174th Meeting, TBD

175th Meeting, Minneapolis, Minnesota, 7–11 May 2018.

FIFTY YEAR AWARDS

"Gold" certificates in recognition of continuing interest in membership in the Society for half a century will be sent to the following members:

C. Allan Boyles
James E. Burke
Malcolm J. Crocker
Donald B. Cruikshank
Lola L. Cuddy
W. Jay Dowling
David Feit
J. Jaycek Figwer
Paul M. Gammell
David A. Harris
Sabih I. Hayek
John F. Hemdal
Milton H. Hieken
Kiyotera Ishii
Jan H. Janssen
Bertram Y. Kinzey, Jr.
Stephen V. Letcher
Jack W. Loney
Albert W. Lowe
Domenic J. Maglieri
William J. Marshall
Robert C. Meyer
James A. Morgan
Giovan G. Natale
David A. Nelson
William H. Perkins
G. Richard Price
David G. Proctor
Lawrence R. Rabiner
James C. Saunders
John T. C. Shen
Toshio Sone
Arnold Starr
Jiri Tichy
Arnie L. Van Buren
Neal F. Viemeister
Richard H. Wilson
Robert S. Winokur

TWENTY-FIVE YEAR AWARDS

The following individuals have been members of the Society continuously for twenty-five years. They will be sent to the following members:

Ronald M. Aarts
Mosato Abe
Joaquim Agullo
Judit Angster
Shigeaki Aoki
Gordon E. Baird
Benjamin A. Bard
Soren Bech
Thomas F. Bergen
Guillaume J. Bock
Arno S. Bommer
Adelbert W. Bronkhorst
Christopher N. Brooks
Shira L. Broschat
James W. Brown
Russell C. Burkhardt
Melanie M. Campbell
Eleanora Carletti
Antonio Pedro O. Carvalho
Pak-kon Choi
Robert L. Clark
John C. Congdon
James P. Cottingham
James T. Cowling
Samuel E. Craig
Richard L. Culver
Diane Dalecki
Allan G. Dallas
Keith Denning
Alan M. Dent
Mumtaz A. Dinno
Felicia M. Doggett
Geoffrey S. Edelson
Joseph J. Eggermont
Kent E. Eschenberg
Carol Y. Espy-Wilson
Stephen A. Fausti
Laurie T. Fialkowski
Vencenzo Gambino
Vladimir Genis
Klaus Genuit
Keith A. Gillis
Joo Thiam Goh
David H. Griesinger
Ellen C. Haas
Babur B. Hadimioglu
Kevin D. Heaney
Michelle L. Hicks
Tatsuya Hirahara
Kullervo H. Hynynen
Toshio Irino
Russell J. Jeffers
Christiaan Kauffman
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Session 1aAA

Architectural Acoustics, Noise and Signal Processing in Acoustics: Spherical Array Processing for Architectural Acoustics

Michael Vorländer, Cochair
ITA, RWTH Aachen University, Neustr. 50, Aachen 52066, Germany

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

Chair's Introduction—8:00

Invited Papers

8:05

1aAA1. Some spatial analysis of the room impulse response. Jens Meyer (mh Acoustics, 38 Meade Rd., Fairfax, VT 05454, jmm@mhacoustics.com) and Gary W. Elko (mh Acoust., Summit, NJ)

The room impulse response (RIR) and measures derived from it have been a powerful tool for room characterization. Traditionally, these measures are based on time domain characteristics like reverberation time or clarity. By recoding a group of RIRs with microphone arrays, one can add the spatial dimension to the RIR. Spherical microphone arrays are of special interest since they can give equal weight to all directions. In this paper, we explore an adaptive cardioid beampattern algorithm for its suitability as a tool for the spatial analysis of RIRs. The algorithm uses the spherical harmonic base patterns ("Eigenbeams") of a spherical microphone array to form and steer a cardioid beampattern. The steering is automatically adjusted to minimize the output power in a specific time region. Due to the cardioid beampatterns sharp null in the beampattern, an adaptive cardioid algorithm can provide detailed spatial information on reflections as a function of time for impulse responses.

8:25

1aAA2. A new technique to measure directional impulse responses using a 64-channel spherical microphone. Jacob Adelgren and David T. Bradley (Phys. + Astronomy, Vassar College, 124 Raymond Ave., Poughkeepsie, NY 12604, jaadelgren@vassar.edu)

A spherical microphone is an array consisting of numerous microphones arranged almost uniformly on the surface of a rigid sphere. Recently, spherical microphone arrays have received attention in architectural acoustics for their ability to characterize the sound field over a full sphere, which can yield additional information about the spatial characteristics of the field, including field diffusivity and directionality of individual reflections. Typically, these arrays are used as real-time analyzers to capture sound pressure information. However, in architectural acoustics, an impulse response measurement is required to adequately characterize the space. An effective approach for utilizing spherical microphones, particularly those with a large number of channels, to capture directional impulse responses has yet to be realized. The current project has utilized a 64-channel spherical microphone to carry out the integrated impulse response measurement technique using a sine-sweep signal. MATLAB has been used to evaluate sound fields in a reverberant chamber using a variety of directionality patterns (e.g., omnidirectional, cardioid, and figure-of-eight). Additionally, work is in progress to allow for arbitrary beamforming and other directionality patterns. Results and analysis will be presented.

8:45

1aAA3. An open-source spherical microphone array design. John Granzow, Tim O'Brien, Darrell Ford, Yoo H. Yeh, Yoomi Hur (CCRMA, Stanford Univ., 660 Lomita Dr., Stanford, CA 94305, granzow@ccrma.stanford.edu), Darius Mostowfi (Sigma Eng., San Carlos, CA), and Jonathan S. Abel (CCRMA, Stanford Univ., Stanford, CA)

We present an open-source design for producing a spherically baffled, 32-channel microphone array having elements aligned with the vertices and face centers of a dodecahedron, and fabricated using additive manufacturing (3-D printing). The design approach emphasized low cost, assembly ease, and acoustic integrity. Mechanical, electrical, and acoustical design considerations are discussed, as are assembly and calibration details. The baffle is a 10-cm-radius sphere, built from separable hemispheres, and supported by a 2-cm-diameter cylindrical stand that serves as a conduit for the microphone signal cables. Microphone capsules are mounted on preamp boards, which provide balanced line level outputs, and press fit to the baffle. The performance of the array is characterized, and an example application to spatial room impulse response measurement is provided. Design documents, including the enclosure model and preamp electrical design and board layout, are provided at https://ccrma.stanford.edu/~granzow/sphericalmicarray/

9:05

1aAA4. Equivalence of plane wave and spherical harmonics rendering of binaural room impulse response. Zamir Ben-Hur, Jonathan Sheaffer, Boaz Rafaely (Elec. & Comput. Eng., Ben-Gurion Univ. of the Negev, Be'er Sheva, Be'er Sheva 8410501, Israel, zami@post.bgu.ac.il)

Binaural technology has various applications in virtual acoustics, architectural acoustics, tele-communications, and auditory science. One key element in binaural technology is the binaural room impulse response (BRIR), which represents a continuum of plane waves spatially filtered by head related transfer functions (HRTFs). Such BRIRs can be rendered from spherical microphone array recordings and free-field HRTFs, either in the space domain using plane-wave composition or in the spherical-harmonics domain using order-limited spherical harmonics representation of the sound field. While these approaches have been individually employed in a number of recent studies, it appears that the literature does not offer a comprehensive analysis or a theoretical framework relating the two representations with respect to binaural reproduction and perception. In this paper, we provide a mathematical analysis showing that when certain sampling conditions are maintained, the plane-wave and spherical-harmonics representations are equivalent. Further, we show that under these conditions, resulting binaural signals are independent of the employed spatial sampling schemes. The analysis is complemented by a listening experiment, in which both plane-wave and spherical-harmonics representations are perceptually evaluated for different spatial sampling schemes and spherical harmonic orders.

9:25

1aAA5. Binaural perception of direct and reverberant sound fields rendered with mixed-order spherical harmonics. Jonathan Sheaffer and Boaz Rafaely (Elec. and Comput. Eng., Ben-Gurion Univ., Beer-Sheva, Beer-Sheva 8410501, Israel, sheaffer@ee.bgu.ac.il)

Binaural responses can be rendered from a plane-wave decomposition of a measured or a modeled sound field, spatially integrated with free-field head-related transfer functions. When represented in the spherical-harmonics domain, the decomposition order reflects the maximum spatial resolution, which is limited by the number of microphones in the spherical array. Recent studies suggest a direct relationship between decomposition order and perceptual attributes such as localization blur, timbre, and the sense of externalization. Insofar, studies have employed plane wave density functions in which the different components of the sound field were uniformly decomposed at a single spherical-harmonics order. This work is concerned with binaural signals in which the direct and reverberant parts of the sound field are decomposed at different spherical-harmonics orders. The direct component of the sound field carries significant directional information utilized by the auditory system. Therefore, changing the spherical-harmonics order of the direct component is expected to have a different perceptual impact compared to changing the spherical-harmonics order of the reverberant part. Listening experiments are employed to study the perception of such mixed-order representations in context of sound localization and auditory distance perception.

9:45–10:00 Break

10:00

1aAA6. Investigation of listener envelopment using spherical microphone array measurements and ambisonics playback. David A. Dick and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dad325@psu.edu)

An important aspect of overall room impression is listener envelopment (LEV), the sense of being immersed in a sound field. Current LEV objective metrics are primarily based on lateral reflections, typically measured with a figure-of-eight microphone. The purpose of this study was to investigate LEV using measured impulse responses (IRs) taken with an Eigenmike 32-element spherical microphone array as an initial step toward creating a new metric for LEV. The spherical array enables a spatial analysis with higher resolution than traditional methods. Impulse response measurements were made in the Peter Kiewit Concert Hall in Omaha, NE, in several seat positions and hall absorption settings. Auralizations were generated by convolving the measured IRs with anechoic music excerpts and then processing the signals for third-order ambisonics playback. The signals were played over an array consisting of 32 loudspeakers in an anechoic chamber. A subjective study was run in which musically trained listeners rated the LEV of the stimuli. Beamforming techniques were used to analyze the IRs to better understand the spatial and temporal factors that contribute to the perception of envelopment. Results will be presented, which correlate the objective measurements to the subjective ratings. [Work supported by NSF Grant 1302741.]

10:20

1aAA7. The effect of playback method on listeners' judgments of concert hall auralizations. Samuel W. Clapp (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Arcisstr. 21, München 80333, Germany, samuel.clapp@tum.de), Anne E. Guthrie (Arup Acoust., New York, NY), Jonas Braasch, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Previous studies of the perception of concert hall acoustics have generally employed one of two methods for soliciting listeners' judgments. The first is to have listeners rate the sound of a hall while physically present in that hall. The second is to record the acoustics of a hall and later simulate those acoustics in a laboratory setting. While the first method offers a completely authentic presentation of the concert experience, the second allows for more direct comparisons between different spaces and affords the researcher greater control over experimental variables. Higher-order spherical microphone arrays offer a way to capture the spatial components of a concert hall's acoustics, which can then be reproduced using a loudspeaker array. In this study, eight different concert and recital halls were measured using both a spherical microphone array and binaural dummy head as receivers. Listeners were then presented with auralizations of the halls using an ambisonic loudspeaker array and headphones, and asked to rate the halls based on subjective preference and on similarity to one another. The responses were analyzed using multidimensional scaling methods in order to examine the effect of the auralization system on the listeners' judgments.

10:40

1aAA8. Spatial analysis of sound fields in rooms using spherical MIMO systems. Hai Morgenstern (Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, POB 653, Beer Sheva 84105, Israel), Markus Noisternig (UMR IRCAM-CNRS-UPMC, Paris, France), and Boaz Rafaely (Elec. and Comput. Eng., Ben-Gurion Univ. of the Negev, Beer Sheva, Israel, br@bgu.ac.il)

The perception of sound by human listeners in a room has been shown to be affected by the spatial attributes of the sound field. These spatial attributes have been studied using microphone and loudspeaker arrays separately. Systems that combine both loudspeaker and microphone arrays, termed multiple-input multiple-output (MIMO) systems, facilitate enhanced spatial analysis compared to systems with a single array, thanks to the simultaneous use of the arrays and the additional spatial diversity. Using MIMO systems, room impulse responses (RIRs) can be presented using matrix notation, which enables a unique study of a sound field's spatial attributes, employing methods from linear algebra. For example, a matrix's rank and null space can be studied to reveal spatial information on a room, such as the number of dominant room reflections and their direction of arrival to the microphone array and the direction of radiation from the loudspeaker array. In this contribution, a theory of the spatial analysis of a sound field using a MIMO system comprised of spherical arrays is developed and a simulation study is presented. In the study, tools proposed for processing MIMO RIRs with the aim of revealing valuable information about acoustic reflections paths are evaluated.

11:00

1aAA9. Practical approach to forward and reciprocal sequential array measurements. Johannes Klein and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, johannes.klein@akustik.rwth-aachen.de)

Room impulse response measurements including directivity are becoming a vital component of room acoustics. The results of these measurements allow for the room acoustical analysis regarding specific source receiver combinations and enable the realistic simulation of acoustical scenarios in virtual environments. Besides the widely applied microphone-arrays, electroacoustic sources with a steerable directivity are necessary to retain all degrees of freedom during the measurements. Due to the relatively large dimensions of the transducers, the number of possible physical transducers in a loudspeaker-array is very limited, resulting in a low spatial resolution. A solution to this problem are sequential measurement methods, which virtually enlarge the number of transducers by moving the source during the measurement and superposing the results. The attainable high spatial resolution is ideal for substituting the most complex source or receiver directivity in forward or reciprocal measurements. However, sequential measurement methods are very susceptible to time variances, which diminish the achievable accuracy. This study is an experimental approach with various source receiver combinations to evaluate the practically achievable accuracy in forward and reciprocal sequential loudspeaker-array measurements in different environments and the corresponding sources of error.

11:20

1aAA10. Spatial aliasing study using a high-resolution radiation measurement of a violin. Noam R. Shabtai, Gottfried Behler, and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, nsh@akustik.rwth-aachen.de)

Sound source radiation patterns can be employed in a virtual acoustic system in order to improve the realistic experience of the listener. Many studies were described in the literature, in which spherical microphone arrays and spherical harmonics were used to capture the radiation pattern of musical instruments. In these studies, however, a limited number of microphones is used due to technical reasons that involve a non-repeatable excitation by the human player and the corresponding requirement to capture the radiation in one single excitation. Up to now, the radiation pattern is measured without a reference that describes the degree of spatial aliasing caused by the limited number of microphones. This work presents a high-resolution spatial sampling of the radiation pattern of an electrically excited violin. An analytical measure to the degree of spatial aliasing is represented and calculated for each number of microphones, using the high-resolution measurement as a reference.

11:40

1aAA11. Array processing methods for the determination of reflection properties of architectural surfaces. Markus Müller-Trapet and Michael Vorländer (Inst. of Tech. Acoust., RWTH Aachen Univ., Kopernikusstr. 5, Aachen 52074, Germany, mmt@akustik.rwth-aachen.de)

It has become popular in architectural acoustics to use microphone arrays, very often in spherical arrangements, to capture and analyze the sound field in rooms. This contribution will present a hemispherical receiver array, designed to obtain information about sound reflection from architectural surfaces, preferably *in-situ*. Similarly to previous studies, the analysis in the spherical harmonics (SH) domain is favored, with the additional challenge of data available on a hemisphere instead of a complete sphere. This problem is solved by obtaining orthonormal base functions on the hemisphere. As application cases, spherical beamforming for the determination of reflection factors will be presented as well as scattering near-field holography in order to determine the diffusion coefficient of small samples. Results from numerical and experimental case studies will be discussed.

Session 1aAB

Animal Bioacoustics: Bioacoustics and Behavior

Samuel L. Denes, Chair
Acoustics, Pennsylvania State Univ., 116 Applied Science Building, University Park, PA 16802

Contributed Papers

8:30

1aAB1. Individually distinctive parameters in the upcall of the North Atlantic right whale (*Eubalaena glacialis*). Jessica A. McCordic and Susan E. Parks (Biology, Syracuse Univ., 114 Life Sci. Complex, Syracuse, NY 13244, jamccord@syr.edu)

According to the source-filter hypothesis proposed for human speech, physical attributes of the vocal production mechanism combine independently to result in individually distinctive vocalizations. In the case of stereotyped calls with all individuals producing a similar frequency contour, the filtering of the signal resulting from the shape and size of the vocal tract may be more likely to contain individually distinctive information than parameters measured from the fundamental frequency resulting from the vibrating source. However, the formant structure resulting from such filtering has been historically undervalued in the majority of studies addressing individual distinctiveness in non-human species. The upcall of the North Atlantic right whale (*Eubalaena glacialis*) is characterized as a stereotyped contact call, and visual inspection of upcall spectrograms confirms presence of a robust formant structure. Here we present preliminary results testing individual distinctiveness of upcalls recorded from archival, suction cup mounted tags (Dtags). Parameters measured from the fundamental frequency contours as well as the formant structure of the calls are used in assigning upcalls to individual whales. These results provide a baseline for further development of acoustic detection techniques that could be used to noninvasively track movements of individual whales across habitats.

8:45

1aAB2. Exploring the vocal ontogeny of North Atlantic right whales. Holly Root-Gutteridge, Dana Cusano (Dept. of Biology, Syracuse Univ., Syracuse, NY 13244, hrootgut@syr.edu), Lisa Conger, Sofie Van Parijs (Northeast Fisheries Sci. Ctr., NOAA Fisheries, Woods Hole, MA), and Susan Parks (Dept. of Biology, Syracuse Univ., Syracuse, NY)

The vocal ontogeny of species can provide insight into their physical and social development and capacity for social learning. North Atlantic right whales (*Eubalaena glacialis*) are an endangered species with a complex range of vocalizations used in a wide range of social contexts. In this study, we systematically characterize developmental changes in sound production from birth through adulthood. Calls have been recorded as part of ongoing North Atlantic right whale research projects spanning 2001 through 2015, and include data from calves as young as 1 month of age. Data included recordings from single hydrophones, hydrophone arrays, and non-invasive digital acoustic recording tags. Only calls which could be confidently attributed to a specific whale of known age were used in the analysis. Developmental periods consisted of 0–3 months, 3–6 months, 6–12 months, and then discrete ages by year. Calls of both male and female calves were included in the analysis. Calls were also classified to age categories using discriminant function analysis (DFA) to determine whether a calf could be aged by its call. Analyses indicate a gradual maturation of sound production with increasing age of individuals.

9:00

1aAB3. Objective analysis of vocal sequences using nested self-organizing maps. Eduardo Mercado (Dept. of Psych., Univ. at Buffalo, Buffalo, NY 14260, emiii@buffalo.edu)

Traditional approaches to analyzing vocal sequences typically involve identifying individual sound units, labeling identified sounds, and describing the regularity of label sequences [A. Kershenbaum *et al*., "Acoustic sequences in non-human animals: A tutorial review and prospectus," Biol. Rev. (2014)]. Although this method can provide useful information about the structure of sound sequences, the criteria for determining when distinct units have been successfully classified are often subjective and the temporal dynamics of sound generation are usually ignored. Self-organizing maps (SOMs) provide an alternative approach to classifying inputs that does not require subjective sorting or isolation of units. For instance, SOMs can be used to classify fixed duration frames sampled from recordings. Once an SOM has been trained to sort frames, the temporal structure of a vocal sequence can be analyzed by training a second SOM to sort spatiotemporal patterns of activation within the frame-sorting SOM. Analyzing humpback whale "song" using this technique revealed that: (1) perceptually warped spectra from frames varied uniformly along several continua; (2) a subset of frame patterns (sound types) was more prevalent; and (3) produced features varied systematically as a function of sequential position within a song for some sounds, but not others.

9:15

1aAB4. Comparison of managed care and wild walrus source characteristics. Samuel L. Denes (Biology, Syracuse Univ., 116 Appl. Sci. Bldg., University Park, Pennsylvania 16802, sld980@psu.edu), Jennifer L. Miksis-Olds (Appl. Res. Lab., The Penn State Univ., University Park, PA), Dave Mellinger (Cooperative Inst. for Marine Resources Studies, Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR), Eric Otjen (Animal Care, SeaWorld San Diego, San Diego, CA), and Ann E. Bowles (Hubbs-SeaWorld Res. Inst., San Diego, CA)

Male Pacific walruses perform acoustic displays while in rut. The purpose of these displays is unknown but are hypothesized to be for territory defense or mate advertisement. Understanding source characteristics will allow the estimation of perceptibility by conspecifics. The displays occur in the Bering Sea in late winter where direct human observation is difficult. Working with an animal in managed care provided the ability to make direct observations of a male producing breeding vocalizations and the direct calculation of source level. Source characteristics from recordings of managed care and wild walruses were analyzed. The mean peak source level of the impulsive knocks produced by the managed care male was 183 dB (re: 1 μPa). The mean peak source level from the wild recordings was 177 dB (re: 1 μPa). For both wild and managed care vocalizations, a significant relationship between ambient noise level and source level was identified. An increase of approximately 5 dB in source level was found for an increase in 10 dB of noise level.

9:30

1aAB5. Bleats are universally used by ungulates for mother-offspring communication but what use do they serve for toads and pandas? David Browning (139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com)

Bleats are usually associated with sheep and goats, but are universally used by ungulates for mother-offspring communication. These short, relatively simple vocalizations are used, depending on circumstance, for localization, identification, emotion, and guidance. Interestingly, there are at least two other cases of bleat users. The Australian bleating toad appears to use them as an efficient means of continuous vocalization, while pandas use bleats as part of a repertoire of jungle talk to communicate in dense bamboo thickets.

9:45

1aAB6. Mobility of dinoflagellates measured by high-frequency ultrasound. Hansoo Kim (Ocean System Eng., Jeju National Univ., Ocean Science College 4, Jeju 690-756, South Korea, hansoo5714@naver.com), Tae-Hoon Bok (Phys., Ryerson Univ., Toronto, ON, Canada), Kweon-Ho Nam, Juho Kim, Dong-Guk Paeng (Ocean System Eng., Jeju National Univ., Jeju, South Korea), So-Jeong An, and Joon-Baek Lee (Earth and Marine Sci., Jeju National Univ., Jeju, Korea (the Republic of))

The importance of phytoplankton contributing more than 50% of the global amount of photosynthesis has been emphasized for a long time. Sometimes, the over-growth of phytoplankton causes the negative influence such as red-tide phenomenon on marine ecological environments. Therefore, the measurement of the mobility of phytoplankton is important. In this study, the mobility of the benthic dinoflagellate, *Amphidinium carterae* Hulburt (*A. Carterae*) incubated by f/2 medium was investigated using high-frequency ultrasound. Backscattering signal from *A. Carterae* was measured for 2 s in every 2 min by a 40-MHz ultrasound transducer, and the integrated backscattering power calculation was followed. The mobility of *A. carterae* in response to the light was illustrated by M-mode image of the echoed signals. The mobility of *A. carterae* was estimated to about 0.4 mm/s for the upward movement in response to light, while its sedimentation rate was measured to about 0.1 mm/s in a dark environment. This study suggests that mobility of benthic dinoflagellates responding to light can be measured by M-mode imaging of high-frequency ultrasound. (This research was a part of the project titled "Measurement of cells division and photosynthesis of phytoplankton using ultrasound", funded by the Ministry of Oceans and Fisheries, Korean.)

MONDAY MORNING, 18 MAY 2015 — BALLROOM 1, 10:00 A.M. TO 1:00 P.M.

Session 1aED

Education in Acoustics: Hands-On Acoustics Demonstrations for Middle- and High-School Students

Cameron T. Vongsawad, Cochair
Physics & Astronomy, Brigham Young University, 1041 E. Briar Avenue, Provo, UT 84604

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

Invited Paper

10:00

1aED1. Auditory illusion demonstrations as related to prehistoric cave art and Stonehenge. Steven J. Waller (Rock Art Acoust., 5415 Lake Murray Blvd. #8, La Mesa, CA 91942, wallersj@yahoo.com)

Auditory illusions will be demonstrated, and their relevance to prehistoric cave art and Stonehenge revealed. In the ancient past, when the wave characteristics of sound were not understood, virtual sound effects arising from complex sound wave interactions (echoes, reverberations, interference patterns, etc.) were misinterpreted as invisible beings (echo spirits, thunder gods, sound absorbing bodies, etc.) as described in ancient myths around the world. In this session, live hands-on demonstrations will be given to small groups of students who can experience for themselves these types of auditory illusions. Participants will get to experience various sounds and will be given the task of interpreting what the sounds are, first blindfolded, then with visual cues. (Previous student reactions have included "Whoa!," "Wow!," "Amazing!," "Cool!," and "What??.") These scientifically conducted experiments show how various ambiguous sounds can be interpreted in more than one way—like optical illusions—and thus can help in understanding our ancestors' reactions to sounds they considered mysterious and spooky. These discoveries are just a few examples of research findings that are springing from the new field of Archaeoacoustics. See https://sites.google.com/site/rockartacoustics/ for further examples.

Session 1aPA

Physical Acoustics: General Topics in Physical Acoustics I

Brian E. Anderson, Chair
Geophysics Group (EES-17), Los Alamos National Laboratory, MS D446, Los Alamos, NM 87545

Contributed Papers

8:30

1aPA1. The vibroacoustical environment in two nuclear reactors. Joshua Hrisko, Steven L. Garrett (Grad. Prog. in Acoust., Penn State, Graduate Program in Acoust., Appl. Res. Lab, State College, PA 16804, jqh5657@psu.edu), Robert W. Smith (Appl. Res. Lab., Penn State, State College, PA), James A. Smith, and Vivek Agarwal (Fundamental Fuel Properties, Idaho National Lab., Idaho Falls, ID)

Laboratory experiments have suggested that thermoacoustic engines can be incorporated within nuclear fuel rods. Such engines would radiate sounds that could be used to measure and acoustically-telemeter information about the operation of the nuclear reactor (e.g., coolant temperature or fluxes of neutrons or other energetic particles) or the physical condition of the nuclear fuel itself (e.g., changes in porosity due to cracking, swelling, evolved gases, and temperature) that are encoded as the frequency and/or amplitude of the radiated sound [IEEE Measurement and Instrumentation **16**(3), 18–25 (2013)]. For such acoustic information to be detectable, it is important to characterize the vibroacoustical environments within reactors. We will present measurements of the background noise spectra (with and without coolant pumps) and reverberation times within the 70,000 gallon pool that cools and shields the fuel in the 1 MW research reactor on Penn State's campus using two hydrophones, a piezoelectric projector, and an accelerometer. Background vibrational measurement taken at the 250 MW Advanced Test Reactor, located at the Idaho National Laboratory, from accelerometers mounted outside the reactor's pressure vessel and on plumbing, will also be presented to determine optimal thermoacoustic frequencies and predict signal-to-noise ratios under operating conditions. [Work supported by the U.S. Department of Energy.]

8:45

1aPA2. Using helium as the working fluid to improve efficiency of high-frequency thermoacoustic engines. Nathaniel Wells (Phys., Utah Valley Univ., Orem, TX) and Bonnie Andersen (Phys., Utah Valley Univ., MS 179, 800 W University Pkwy, Orem, UT 84057, bonniem@uvu.edu)

Previous work on thermoacoustic engines with bottle-shaped resonators has been done to improve performance by varying geometric parameters, using air as the working fluid. This study is focused on transitioning from air to helium for the working fluid to further improve device performance. The theoretical ratio of efficiencies was derived for the two operating fluids. The existing engine was redesigned for evacuating the air and introducing helium into the resonator and six different types of heat shrink tubing used to hold the heat exchangers in place were tested for effectiveness with a vacuum and ease of removal. The optimal stack masses for this engine operating with air and helium were theoretically estimated and tested with air using six different stack masses from 50 to 62 mg. The resonator had a cavity with a length of 10 cm and ID of 4.13 cm and a neck with a length of 2.62 cm and ID of 1.91 cm and used steel wool for the stack material. The engine was supplied 12 W from a heating element applied above the hot heat exchanger in the neck. The engine was allowed to run for 40 s after the temperature had reached a steady state and the acoustic pressure at the bottom of the cavity was measured. The optimal amount of stack in air was found to be 56 mg, and the acoustic pressure was 206 Pa, Pk-Pk. The adhesive heat shrink tubing was found to be the most effective for use with helium and ease of removal.

9:00

1aPA3. Differences in atmospheric absorption coefficients between ANSI/ASA S1.26-2014 and an updated model. Erik A. Petersen and Victor W. Sparrow (Graduate Program in Acoust., Penn State Univ., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16801, eap206@psu.edu)

Absorption coefficient predictions from the ANSI/ASA standard S1.26-2014 and Sutherland and Bass 2004 [Sutherland, *et al*., J. Acoust. Soc. Am. **115**(3), 2004] are compared for 125, 250, 500, and 1000 Hz pure tones ranging in elevation from 0 to 10 km. The differences in absorption mechanisms as well as the assumed atmospheric profiles are considered. Calculated using their respective profiles, the two models differ by 1–4 dB/km. Additionally, cumulative absorption over a 10 km vertical path is calculated under several conditions; ANSI with ANSI profile, ANSI with S&B profile, and S&B with S&B profile. Comparing the second and third case shows a difference of 0.5–3 dB, which characterizes model dependent differences by evaluating both models using the same profile. A comparison of the first and second cases yields a difference of 0.5 to 10 dB, indicating a strong dependence on the assumed atmospheric profile. To achieve consistent predictions over a 10 km path the layer thickness should be no greater than 1 km. [The opinions, findings, conclusions, and recommendations expressed here are those of the authors and do not necessarily reflect the views of sponsors of the ASCENT Center of Excellence including the Federal Aviation Administration.]

9:15

1aPA4. High frequency oblique-angle acoustic reflections from an air-snow interface. Donald G. Albert, Arnold J. Song, and Zoe R. Courville (ERDC-CRREL, 72 Lyme Rd., Hanover, NH 03755, donald.g.albert@usace.army.mil)

Fresh natural snow is a difficult material to characterize, as any mechanical interaction is likely to damage the fragile pores and grain bonds. Because acoustic waves are sensitive to the porous material properties, they potentially can be used to measure snow properties in a non-destructive manner. Such methods have already been demonstrated on cohesive porous materials including manufactured foams, porous metals, and sintered glass beads. As a first step toward developing a portable method that can be used outdoors, we conducted high frequency oblique-angle reflection measurements on snow samples in a cold room. We compare the acoustically derived parameters with microcomputerized tomography (CT) methods and with standard (but destructive) laboratory measurements. [This research funded by the U.S. Army Corps of Engineers.]

9:30

1aPA5. Comparisons between physics-based, engineering, and statistical learning models for outdoor sound propagation. Nathan J. Reznicek, Carl R. Hart, D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, carl.r.hart@usace.army.mil), Chris L. Pettit (Aerosp. Eng. Dept., U.S. Naval Acad., Annapolis, MD), and Edward T. Nykaza (U.S. Army Engineer Res. and Development Ctr., Champaign, IL)

Many outdoor sound propagation models exist, ranging from highly complex physics-based simulations to simplified engineering calculations. More recently, highly flexible statistical methods have become available, which offer the capability to predict outdoor sound propagation, given a training dataset. Among the variety of modeling approaches, the range of incorporated physics varies from none, as in the statistical learning methods, to comprehensive considerations for physical models. Engineering methods vary in the level of incorporated physics, oftentimes resorting to heuristic or approximate approaches. In order to compare the capability of engineering and statistical learning models, one particular physics-based model is used for outdoor sound propagation predictions, namely, a Crank-Nicholson parabolic equation (CNPE) model. Narrowband transmission loss values predicted with the CNPE, based upon a simulated dataset of meteorological, boundary, and source conditions, act as simulated observations. Among the engineering models used in the comparisons are the Harmonoise propagation model and the ISO 9613-2 method. Among the statistical learning methods used in the comparisons is a random forest regression model. Metrics such as the root-mean-square error and the skill score are computed for both the engineering models and statistical learning models.

9:45

1aPA6. Use of evanescent plane waves for low-frequency energy transmission across material interfaces. Daniel C. Woods, J. S. Bolton, and Jeffrey F. Rhoads (School of Mech. Eng., Purdue Univ., 585 Purdue Mall, School of Mech. Eng., West Lafayette, IN 47907, woods41@purdue.edu)

The transmission of sound across high-impedance difference interfaces, such as an air-water interface, is of significant interest for a number of applications. Sonic booms, for instance, may affect marine life, if incident on the ocean surface, or impact the integrity of existing structures, if incident on the ground surface. Reflection and refraction at the material interface, and the critical angle criteria, generally limit energy transmission into higher-impedance materials. However, in contrast with classical propagating waves, spatially decaying incident waves may transmit energy beyond the critical angle. The inclusion of a decaying component in the incident trace wavenumber yields a nonzero propagating component of the transmitted surface normal wavenumber, so energy propagates below the interface for all oblique incident angles. With the goal of investigating energy transmission using incident evanescent waves, a model for transmission across fluid-fluid and fluid-solid interfaces has been developed. Numerical results are shown for the air-water interface and for common air-solid interfaces. The effects of the incident wave parameters and interface material properties are also considered. For the air-solid interfaces, conditions can be found such that no reflected wave is generated, due to impedance matching among the incident and transmitted waves, which yields significant transmission increases over classical incident waves.

10:00–10:15 Break

10:15

1aPA7. Crank-Nicholson solution of the wide-angle parabolic equation for inhomogeneous moving media. D. K. Wilson and Vladimir E. Ostashev (U.S. Army Cold Regions Res. and Eng. Lab., Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil)

Solutions of the narrow-angle parabolic equation (PE) are essentially the same in non-moving and moving media, due to the validity of the effective sound-speed approximation. However, the wide-angle PE (PE) and its numerical solution are considerably more complicated for a moving than for a non-moving medium. Starting with a rigorously derived, wide-angle PE for propagation in an inhomogeneous moving medium, the Crank-Nicholson solution of this equation is shown to involve a pentadiagonal matrix, which is relatively inefficient to calculate. However, a high-frequency approximation of this equation, valid when the wavelength is small compared to the length scale of the inhomogeneities, leads to a particularly simple wide-angle PE, the solution of which involves a tridiagonal matrix and thus can be solved with only slight extensions to existing narrow-angle PE codes. Solutions of the wide-angle PE are illustrated with examples for sound propagation in the atmosphere. Comparisons to the narrow-angle PE exhibit close agreement at low propagation angles. At higher propagation angles, the refraction effects are found to be relatively unimportant.

10:30

1aPA8. Outdoor measurements of shock-wave propagation from exploding balloons. Sarah M. Young (Dept. of Phys., Brigham Young University-Idaho, Romney 118, Rexburg, ID 83460, sarahmyoung24@gmail.com), Kent L. Gee, Traclanne B. Neilsen, and Kevin M. Leete (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Previous anechoic measurements of exploding latex balloons filled with stoichiometric mixes of acetylene and oxygen revealed how these sources could be used to study weak-shock decay over relatively short ranges [M. B. Muhlestein *et al.*, J. Acoust. Soc. Am. 131, 2422–2430 (2012)]. This paper describes an experiment conducted using a larger balloon over much longer propagation ranges at the Bonneville Salt Flats, which represents a hard, flat, relatively homogeneous ground surface. Measurements of a 0.56 m balloon were made along a propagation radial from 0.31 m from the balloon surface to 1600 m. Data were collected at a sampling rate of 204.8 kHz using piezoresistive pressure gauges and Type-1 condenser microphones. Described are waveform and spectral characteristics, as well as comparisons of the peak pressure decay with the weak-shock model employed previously. Waveform inspection and the comparison indicate that weak shocks are present out to at least 305 m and the amplitude decay rate can be predicted reasonably well using the model. Deviations from the model may be evidence of Mach-like reflections [K. M. Leete *et al.*, Four Corners Ann. Meet. Am. Phys. Soc. 59, I7.00007 (2014)]. This work extends the previous laboratory experiments and serves as a foundation for further studies using this relatively low-cost source.

10:45

1aPA9. Mach reflections in the propagation of outdoor acoustic shocks generated by exploding balloons. Kevin M. Leete, Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, KML@byu.edu), Sarah M. Young (Dept. of Phys. and Astronomy, Brigham Young Univ., Rexburg, Idaho), Tadd T. Truscott, and Jonathon R. Pendlebury (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT)

When a shock wave reflects off a rigid surface, for certain combinations of shock strength and incident angle to the surface a Mach reflection can occur. This is when the incident and reflected shock waves merge to create a stronger shock wave (called a Mach stem) that travels parallel to the reflecting surface and whose height grows with distance. This phenomenon has been studied extensively for large explosions and for steady shock waves, but is less understood for acoustic weak shocks, where current models for Mach stem formation and growth do not agree with experimental observations. A weak-shock propagation experiment has been conducted at the Bonneville Salt Flats using blast waves generated by acetylene-oxygen filled balloons located at a fixed height above the ground. Analysis of acoustic data at various distances from the source and high-speed camera footage both identify a merging of the direct and ground-reflected waves and the formation of a Mach stem at locations closer to the source than theory would otherwise predict.

11:00

1aPA10. Quantitative analysis of a frequency-domain nonlinearity indicator. Kyle G. Miller (Brigham Young Univ., 323 East 1910 South, Orem, UT 84058, millerthepillar@gmail.com), Brent O. Reichman, Kent L. Gee, and Tracianne B. Neilsen (Brigham Young Univ., Provo, UT)

An ensemble-averaged, frequency-domain version of the Burgers equation can be rearranged in order to directly compare the effects of nonlinearity on the sound pressure level, with the effects of atmospheric absorption and geometric spreading on a decibel scale. This nonlinear effect is calculated using the quadspectrum of the pressure and the pressure squared waveforms.

A number of nonlinearity indicators have been developed based on this quadspectral term, but their use has been largely qualitative. Further analysis of the expression has resulted in a quantitative understanding of how a quadspectrum normalization, referred to as Q/S, can be tied to the frequency-dependent sound pressure level changes due to nonlinearity. This understanding is developed by analyzing how Q/S evolves for analytical scenarios—the Blackstock Bridging Function and the Mendousse Solution. The decibel change per shock formation distance calculated through Q/S accurately captures the growth and decay of the higher harmonics, indicating that the most significant changes in the normalized quadspectrum occur before sawtooth formation.

11:15

1aPA11. Underwater laser acoustic source control using shaped plasmas. Theodore G. Jones, Michael Helle, Dmitri Kaganovich, Antonio Ting (Plasma Phys., U.S. Naval Res. Lab., NRL Code 6795, 4555 Overlook Ave. SW, Washington, DC 20375, ted.jones@nrl.navy.mil), Michael Nicholas, David Calvo (Acoust., U.S. Naval Res. Lab., Washington, DC), Gregory DiComo, and James Caron (Res. Support Instruments, Inc., Lanham, MD)

NRL is developing an intense laser acoustic source using underwater shaped plasmas. Recent experiments include near-field acoustic source characterization using high energy lens-focused pulses of a Q-switched Nd:YAG 532 nm laser. The laser-generated plasma evolves into a piston expanding at supersonic speed, which launches an intense shock in the near field. The size and shape of this super-heated piston determines the acoustic waveform and energy spectral density (ESD). We have demonstrated the ability to change the ESD centroid from 15 kHz to a few MHz, with lower frequencies generated using highly elongated plasmas generated by a single laser pulse. We will discuss ongoing laser acoustic source experiments and research plans at NRL involving shaped underwater plasmas, including both demonstrated single-laser-pulse techniques and proposed two-laser-pulse techniques (T. G. Jones, *et al.*, "Two laser generation of extended underwater plasma," U.S. patent application 13/711,752). Two-laser-pulse acoustic generation hold promise for creating meter-scale plasmas, thereby lowering the acoustic ESD to the few-kHz frequency range, which is useful for long-range applications including sonar and communications. Acoustic source characterization includes acoustic waveform and directivity measurements using hydrophones sensitive from 1 Hz to 15 MHz. [This work was supported by NRL Base Funds.]

11:30

1aPA12. Chemical kinetics theory of pyrotechnic whistles. Gregory W. Lyons and Richard Raspet (National Ctr. for Physical Acoust., The Univ. of MS, NCPA, P.O. Box 1848, University, MS 38677-1848, gwlyons@go.olemiss.edu)

Pyrotechnic whistles are sound effect devices commonly used in fireworks and consist of a particular fuel-oxidizer mixture pressed into the bottom of a tube. When ignited, a loud sound is emitted with harmonic frequencies corresponding to standing-wave modes of the tube. A theory of pyrotechnic whistles is developed for a chemical kinetics feedback mechanism in a model combustion reaction. Normal modes are obtained for a cylinder with steady, uniform axial flow. Boundary conditions are derived for a state-dependent reaction rate in an infinitesimal combustion surface. Solutions are presented for the normal modes with respect to reaction rate and tube outlet impedance.

MONDAY MORNING, 18 MAY 2015 — KINGS 3, 8:00 A.M. TO 12:00 NOON

Session 1aPPa

Psychological and Physiological Acoustics and Animal Bioacoustics: Kinematic Hearing: Auditory Perception of Moving Sounds by Moving Listeners

W O. Brimijoin, Cochair
Institute of Hearing Research, Medical Research Council, 10-16 Alexandra Parade, Glasgow G31 2ER, United Kingdom

Michael Akeroyd, Cochair
MRC/CSO Institute of Hearing Research - Scottish Section, New Lister Building, Glasgow Royal Infirmary, Glasgow G31 2ER, United Kingdom

Cynthia F. Moss, Cochair
Psychological and Brain Sci., Johns Hopkins Univ., 3400 N. Charles St., Ames Hall 200B, Baltimore, MD 21218

Chair's Introduction—8:00

Invited Papers

8:05

1aPPa1. Rotating sound sources and listeners: Sound source localization is a multisensory/cognitive process. William Yost, Xuan Zhong, and Anbar Najam (Speech and Hearing Sci., ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

When sound sources or listeners rotate, the acoustic cues used for sound source localization change, but in the everyday world listeners perceive sound rotation only when the sound source rotates not when the listener rotates. That is, in the everyday world, sound source locations are referenced to positions in the environment (a world-centric reference system). The acoustic cues for sound source location

indicate a sound source's location relative to the head (a head-centric reference system), not locations relative to the world. To compute world-centric locations of sound sources, the auditory spatial system must have information about the acoustic cues used for sound source location and cues about the position of the head. The use of visual and vestibular information in determining head position in sound rotation perception was investigated in four experiments. The experiments clearly show, for the first time, that sound source localization when sound sources and listeners rotate is based on acoustic and visual, and sometimes vestibular information. Sound source localization is not based just on acoustics. It is a multisensory process. [Research supported by an AFOSR grant.]

8:25

1aPPa2. Updating and orientation in auditory space. Daria Genzel (Div. of Neurobiology, Dept. Biology II, Univ. of Munich, Grosshaderner Str. 2, Munich 82152, Germany), Paul MacNeilage (German Ctr. for Vertigo and Balance Disord., University Hospital of Munich, Munich, Germany), and Lutz Wiegrebe (Div. of Neurobiology, Dept. Biology II, Univ. of Munich, Munich, Germany, lutzw@lmu.de)

For stable sound localization in space, a listener has to account for self-motion. This requires a head-to-world coordinate transformation by combining proprioceptive and vestibular inputs with the binaural cues for sound localization. Proprioceptive and vestibular information influence the evaluation of spatial auditory cues as indicated by the numerous proprioceptive and vestibular neural inputs into the auditory brainstem. First, we evaluate the relative influence of vestibular and proprioceptive cues on updating the world-centered position of auditory targets during active and passive head and/or body rotations in azimuth. Our results show that both vestibular and proprioceptive signals are used to update the spatial representations of auditory targets, but that vestibular inputs contribute more than proprioceptive inputs. Second, we explore the interplay of spatial audition and self-motion in subjects for which auditory space evaluation is of utmost importance, namely, for blind humans relying on echolocation. Again, the techniques and paradigm allow disentangling vestibular and proprioceptive components for effective orientation based on the auditory analysis of the echoes of self-generated sounds. Our data show how good human biosonar can get for spatial orientation and how self motion helps suppressing orientation biases by lateral walls and front-back confusions.

8:45

1aPPa3. Hearing motion in motion. Simon Carlile, Johahn Leung, Shannon Locke, and Martin Burgess (School of Medical Sci., Univ. of Sydney, F13, Anderson Stuart Bldg., Camperdown, Sydney, New South Wales 2006, Australia, simonc@physiol.usyd.edu.au)

The acoustic cues to auditory space are referenced to the head which moves through the world, itself composed of moving sources so that our sensation convolves source with self- motion. With the head still, velocity discrimination, a perceptual process, is related to static acuity via the minimum audible movement angle (MAMA). Yet, while we can accurately localize static sounds, we are much less sensitive to velocity, resorting to distance and time cues where available. Interestingly, when velocity changes as a step function, discrimination thresholds and the amount of post-transition stimulus required for detection is greater than the corresponding MAMA. This suggests that when the head is stationary, the window of temporal integration may vary according to the sound's velocity characteristics. We also have evidence that auditory representational momentum scales with velocity, not duration or distance. Facing and following a moving auditory source is an ecologically important behavior. Tracking exhibits on-line velocity correction for slow to moderate velocities ($< 80\,^{\circ}$/s) but at higher velocities reflects a more predictive mechanism. Patients with schizophrenia are impaired in their ability to track a moving auditory target, when compared with controls, despite having normal velocity perception when the head is not moving. The presence of other efference copy dysfunctions in schizophrenia suggests a key role for motor efference copy in the disambiguation of self and target motion.

9:05

1aPPa4. Smooth pursuit eye and gaze movements to moving auditory targets: Evidence for velocity-specific processing in the auditory system. Christina Cloninger, Justin T. Fleming, Paul D. Allen, William E. O'Neill, and Gary D. Paige (Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, christina_cloninger@urmc.rochester.edu)

Auditory motion perception remains poorly understood, in contrast with its well-established visual counterpart. Visual smooth pursuit (SP), a velocity-specific behavior, has been well-quantified during both head-fixed (*ocular* SP) and head-free (eye + head, or *gaze* SP) conditions. In contrast, auditory SP tracking has received little attention, despite its potential for demonstrating a motion-specific process in audition. We presented constant-velocity (10–40°/s), free-field auditory (0.2–20 kHz white noise), and visual (LED) targets to head-fixed or head-free subjects while recording *ocular* and *gaze* SP responses, respectively. To control for possible SP in the absence of a target, subjects were asked to recreate auditory trajectories after priming with a set of auditory ramps (practiced SP). We found that *ocular* auditory SP is consistently higher in gain than practiced SP, but variable and lower than visual SP. Further, the gain of auditory *gaze* SP exceeds *ocular* auditory SP. Finally, SP of periodic (triangular) motion trajectories revealed that auditory, like visual, SP improves rapidly over time, indicating predictive behavior. In sum, auditory SP closely mimics visual SP but with reduced gain. We propose that auditory motion processing exists, is robust, and recruits a velocity-dependent neural process shared with vision.

9:25

1aPPa5. Perceived auditory motion is inaccurate during smooth head rotation. Tom C. Freeman, John F. Culling (Psych., Cardiff Univ., Tower Bldg., Park Pl., Cardiff, South Glamorgan CF10 3AT, United Kingdom, freemant@cardiff.ac.uk), Michael A. Akeroyd, and W Owen Brimijoin (Glasgow Royal Infirmary, MRC/CSO Inst. of Hearing Res. (Scottish Section), Glasgow, United Kingdom)

Hearing is confronted by a similar problem to vision as the observer moves. Movement of the sensors creates image motion that remains ambiguous until the observer knows the velocity of eye and head. The visual system solves this problem using motor commands, proprioception, and vestibular information (so-called "extra-retinal signals"), but the solution is not always perfect. Here, we compare the auditory errors made during head rotation with the visual mistakes made during eye movement. Real-time measurements of head velocity were used to change the gain relating head movement to source movement across a loudspeaker array. The gain at which "extra-cochlear signals" (encoding head rotation) was perceptually matched to "acoustic signals" (encoding source motion across the ears), thus yielding the perception of a stationary source, was small and positive. The gain varied depending on context, e.g., average source direction with respect to head. Two possible accounts of analogous findings in vision will be discussed, one based on differences in neural signal accuracy and the other based on Bayesian estimates that resolve differences in neural signal precision. We consider the degree to which these explanations can be applied to auditory motion perception in moving listeners.

9:45–10:00 Break

10:00

1aPPa6. From a nonuniform brain map to motion selectivity in the owl's midbrain. Jose L. Pena (Neurosci., Albert Einstein College of Medicine, 1410 Pelham Pkwy South, Kennedy Ctr. Rm, 529, Bronx, NY 10461, jose.pena@einstein.yu.edu), Brian J. Fischer (Mathematics, Seattle Univ., Seattle, WA), and Yoram Gutfreund (Physiol. and Biophys., Technion Med. School, Haifa, Israel)

The owl's midbrain displays a map of auditory space. This map is a distorted representation of the environment, where the front is magnified. In addition, sound is differently attenuated by the head depending on direction, where gain increases in the front. Because neurons in the map are functionally interconnected, the nonuniform representation influences the processing of features that rely on integration across space and time. In addition, the nonuniform gain deforms spatial receptive fields, affecting history-dependent responses. As a result, neurons become sensitive to motion direction. Previous work has explained the owl's localizing behavior by statistical inference, where uncertainty about the sensory input and prior information can be combined optimally to guide behavior. This theory can be applied to moving targets, where sensory cues must be integrated over time. This analysis shows that the midbrain neural population can be readout to predict future positions of moving targets, a critical function for a predator species. Thus, the nonuniform representation of space can induce biased computation of a higher-order stimulus feature, allowing systematic direction-selectivity and predictive power. Because neural representations where ethologically important ranges are overrepresented, are widespread in the brain, these mechanisms are likely observed in other sensory maps that guide behavior.

10:20

1aPPa7. Detection and tracking of fluttering moths by echolocating bats. Wu-Jung Lee and Cynthia F. Moss (Psychol. and Brain Sci., Johns Hopkins Univ., 3400 N Charles St., Ames Hall 132, Baltimore, MD 21218, wjlee@jhu.edu)

Aerial-hawking echolocating bats present an interesting model for studying how a moving listener interacts with moving sound sources. During foraging, relative movements between the bat and its prey introduce echo variation, which is further influenced by the timing of biosonar emissions with respect to insect wingbeats. Through systematic characterization of echoes from fluttering moths, this study aims at understanding how intermittent and highly variable echo information may be perceived and integrated by foraging bats for prey detection, tracking, and discrimination. Calibrated broadband echoes from frequency-modulated linear sonar sweeps were measured from live, fluttering moths of different morphological features with concurrent high-speed video recordings. The measurements are used to estimate the probability distribution of echo amplitudes as a function of moth morphology, wingbeat phase, and body orientation, as well as to predict changes of echo characteristics as the bat approaches the prey. Signal detection theory is then applied to model the bat's perception of prey presence as influenced by changes in biosonar emission rate in different phases of the foraging process. These results can be further integrated with neurophysiological findings on the bat's internal representation of space for a better understanding of this agile and efficient biosonar system.

10:40

1aPPa8. Navigating the world using echo flow patterns. Michaela Warnecke and Cynthia F. Moss (Dept. of Psych and Brain Sci., Johns Hopkins Univ., 3400 N Charles St., Baltimore, MD 21218, warnecke@jhu.edu)

As an animal moves through the environment, it experiences flow of sensory information from stationary objects. Animals that rely largely on visual information to guide their movement experience optic flow patterns as they navigate, which can be used to measure the relative distance of objects in the environment (Gibson, 1958, Besl, 1988). For example, honeybees use optic flow patterns to center themselves in a flight corridor, and experimental manipulations of the visual patterns on the walls directly influence the animal's navigation path (Srinivasan *et al*., 1996). Other animals instead show wall-following behavior, choosing to navigate close to visual obstacles (Scholtyssek *et al*., 2014). Here, we report on the navigation paths of animals that rely on acoustic signals to guide their movement. Echolocating bats emit ultrasonic signals that reflect from objects in the path of their sound beam, and we are studying how these animals use echo flow to guide their flight path through a corridor. In this study, we flew echolocating big brown bats through tunnels of horizontally and vertically hung PVC pipes to investigate how acoustic patterns influence the bat's flight and echolocation behavior.

11:00

1aPPa9. Aurally aided visual search performance for stationary and walking listeners. Douglas Brungart, Tricia Kwiatkowski (Walter Reed NMMC, 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.brungart@us.army.mil), Sarah E. Kruger (National Intrepid Ctr. of Excellence, Bethesda, MD), Julie Cohen, Thomas A. Heil, and Danielle Zion (Walter Reed NMMC, Bethesda, MD)

One of the most important functions of spatial hearing is to facilitate the rapid shift of visual gaze in the direction of newly detected sound sources that might represent potential targets or threats in our environment. Some of the most dangerous threats can emerge when listeners are walking, so it is useful to know what impact walking might have on audiovisual target detection and identification in complex scenes. This study used a virtual environment consisting of a motorized treadmill surrounded by a 180^{o} projection screen mounted in front of a 64-loudspeaker array to evaluate how quickly participants were able to detect a visual target displayed in the presence of multiple distracters. The distracters were small clusters containing 2 or 4 dots, and the target was a small cluster containing 1 or 3 dots that was presented from the same location as a pulsed broadband noise. The task was to find the target as quickly as possible and press a button to indicate if it had 1 or 3 dots. Data were also collected in an auditory localization condition that required listeners to move a cursor to the location of the sound source and a visual-only condition that required participants to perform the visual search task with no auditory cue. The results show that target identification times were generally reduced when the listener was walking, suggesting that the increased motor activation caused by walking may enhance the ability to perform audiovisual searches. [Research Supported by DoD PHTBI award (W81XWH-12-2-0068).]

11:20

1aPPa10. On the role of visual information about head motion in the interpretation of dynamic interaural cues for front/back sound localization. Ewan A. Macpherson (National Ctr. for Audiol. & School of Communication Sci. and Disord., Western Univ., Elborn College, 2262, 1201 Western Rd., London, Ontario N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

The dynamic interaural cues generated by listener head rotation specify the front/back location of a sound source only when coupled with information about the head motion that produced them. Potential sources of this information are vestibular, proprioceptive, and visual inputs. To investigate the influence of these extra-auditory modalities on the interpretation of dynamic acoustic cues, we use real-time motion tracking and dynamic virtual auditory space synthesis in conjunction with an oscillating chair apparatus that permits dissociation of head-on-body and head-in-space motion. The results of a previous study conducted in darkness [Kim, Barnett-Cowan, & Macpherson, ICA 2013], suggest that vestibular (head-in-space) information is necessary and sufficient for accurate interpretation of dynamic cues, whereas proprioceptive (head-on-body) information is neither necessary nor sufficient. In the present study, active (providing vestibular and proprioceptive information) or passive (providing primarily vestibular information) head-in-space rotations at 50°/s were combined with congruent or incongruent visual motion information in a task requiring the identification of the front/back location of 200- or 400-ms low-pass noise targets. Experiments were conducted in a lighted room, and incongruent visual stimuli were produced with left/right-reversing prism glasses providing ~30° field of view. We found that this visual input had little or no influence on listeners' interpretation of the dynamic auditory cues.

11:40

1aPPa11. Changes in the integration of self motion and auditory spatial cues with age, hearing impairment, and use of hearing devices. W. O. Brimijoin and Micheal A. Akeroyd (Inst. of Hearing Res. - Scottish Section, MRC/CSO, MRC/CSO Inst. of Hearing Res., 10-16 Alexandra Parade, Glasgow G31 2ER, United Kingdom, owen@ihr.gla.ac.uk)

To resolve a front/back confusion, listeners may use both the spectral cues associated with the pinnae and they may turn their heads and note the direction in which the signal moves. Since hearing impairment typically involves the loss of high frequency information, one might expect that hearing impaired listeners would be more reliant on self-motion cues. Such listeners, however, often wear hearing aids that alter binaural level cues, suggesting they may distort dynamic self-motion-related cues. To examine these interactions, we utilized a previously published front/back illusion [W.O. Brimijoin and M.A. Akeroyd, iPercept. 3(3), 179–182 (2012)]: the perceptual location of signals whose position is moved at twice the angular rate of head movements is opposite to its physical location. In normal-hearing listeners, the illusion is powerful but weakens with increasing low-pass filter cutoff frequency. We found that for hearing-impaired listeners, the illusion was effective at all cutoff frequencies. The effect of hearing aids was heterogeneous across listeners, but in no case was a listener returned to normal performance, suggesting that hearing aids are not only failing to provide the listener with spatially-informative spectral cues, but they may interfere with self-motion-related cues. [Work supported by MRC (U135097131) and the Chief Scientist Office (Scotland).]

MONDAY MORNING, 18 MAY 2015 — BALLROOM 2, 8:00 A.M. TO 12:00 NOON

Session 1aPPb

Psychological and Physiological Acoustics: General Topics in Psychological Acoustics (Poster Session)

Nirmal Kumar Srinivasan, Chair
National Center for Rehabilitative Auditory Research, 3710 SW US Veterans Hospital Road, Portland, OR 97239

Posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to view other posters, contributors of odd-numbered papers will be at their posters from 8:15 a.m. to 9:45 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 11:30 a.m. There will be a 15-minute break from 9:45 a.m. to 10:00 a.m.

Contributed Papers

1aPPb1. The three-dimensional morphological database for spatial hearing research of the BiLi project. Felipe Rugeles Ospina (Orange Labs, 4 rue du Clos Courtel, Cesson Sevigné 35510, France, felipe.rugelesospina@orange.com), Marc Emerit (Orange Labs, Cesson Sévigné, France), and Brian F. Katz (LIMSI, Orsay, France)

The BiLi Project is a French collaborative project concerned with binaural listening and improved rendering for the general public. One of the goals of this project is to develop simple methods to personalize head-related transfer functions based on people's morphology. In order to accomplish this, it is necessary to study the links between an individual's HRTF and their corresponding morphology. As a resource for studies relating these parameters, two databases have been created: a database of high resolution measured HRTFs and an accompanying database of 3-D morphological data. This work presents the details of the creation of the morphological database. The prerequisites for such a database are presented. Various technical solutions are proposed and evaluated. Resulting accuracies of the methods are compared using extracted morphological parameters as defined in the CIPIC database with those measured directly on the individuals.

1aPPb2. On the importance of information-bearing acoustic changes for understanding speech in simulated electrical-acoustic stimulation. Christian Stilp (Dept. of Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu), Gail Donaldson, Soohee Oh (Univ. of South Florida, Tampa, FL), and Ying-Yee Kong (Northeastern Univ., Boston, MA)

Listeners utilize information-bearing acoustic changes in the speech signal to understand sentences. This has been demonstrated in full-spectrum speech using cochlea-scaled entropy (CSE; Stilp & Kluender, 2010 *PNAS*) and in vocoded speech (CSE_{CI}; Stilp *et al.*, 2013 *JASA*). In simulations of electrical-acoustic stimulation (EAS), vocoded speech intelligibility is aided by the preservation of low-frequency acoustic cues. The extent to which listeners rely on information-bearing acoustic changes to understand EAS speech is unclear. Here, normal-hearing listeners were presented noise-vocoded sentences with 3–6 spectral channels in two conditions: 1) vocoder-only (80–8000 Hz, filtered using third-order elliptical filters), and 2) simulated hybrid EAS (vocoded >500 Hz; original acoustic <500 Hz). In each sentence, four 80-ms intervals containing high-CSE_{CI} or low-CSE_{CI} acoustic changes were replaced with speech-shaped noise. As expected, performance improved with more channels and the preservation of low-frequency fine-structure cues (EAS). Relative to control vocoded sentences with no noise replacement, performance was impaired more when high-CSE_{CI} intervals were replaced by noise than when low-CSE_{CI} intervals were replaced in 5- and 6-channel sentences, but not at lower spectral resolutions. This effect maintained across vocoder-only and EAS sentences. Findings support the conclusion that EAS users make use of information-bearing acoustic changes to understand speech.

1aPPb3. A study on sound contents development based on analysis a Foley sound and a real sound of thunder. Ahn Iksoo (TeleCommun. & Information, soolsil Univ., 369 sangdo-ro, Dongjak-gu, Seoul 156-743, South Korea, aisbestman@naver.com), Bae Seong Geon (daelim, Anyang, South Korea), and Bae Myungjin (TeleCommun. & Information, soolsil Univ., Seoul, South Korea)

This study focuses on verifying possibility of developing Foley sound of thunder, one of the tools of Foley sound used to make sound effect required for story making of radio drama in early period broadcasting as a sound content by examining it. The purpose of this research is to make its creativity and uniqueness into sound content by proving similarities between thunder Foley sound made by tools and actual Foley sound based on their comparison and analysis and studying its production principles and usage.

1aPPb4. Predicting the timing of future events using sound: Bouncing balls and tone bursts. Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 600 S. Paulina Str., AAC 1012, Chicago, IL 60612, valeriy_shafiro@rush.edu), Brian Gygi (Speech and Hearing Res., U.S. Dept. of Veterans Affairs Northern California Health Care System, Martinez, CA), and Anatoliy Kharkhurin (Dept. of Int.. Studies, American Univ. of Sharjah, Sharjah, United Arab Emirates)

Listeners can predict the timing of the next bounce of a bouncing ball by sound alone with high accuracy, relying primarily on temporal cues [Giordano *et al.*, 2011, JASA, 129, 2594]. The present study investigated the role of temporal structure (natural bouncing patterns versus artificially reversed bouncing patterns) and event type (ball bouncing sounds versus similarly patterned tone bursts). After listening to two to four such sounds, listeners would indicate when they expected the next sound to occur (without hearing it). Results replicate previous findings of high accuracy in predicting the timing of natural bounce patterns regardless of event type. In contrast, accuracy was substantially poorer when pattern timing was reversed. Nevertheless, even for reversed patterns, listener accuracy improved as a greater number of bounces were heard prior to response time. This suggests that with additional information listeners were able to utilize veridical acoustic cues that best fit the temporal pattern. These findings demonstrate that in addition to being highly adept in estimating future timing of natural auditory events, listeners (a) tend to rely on temporal dynamics of everyday events in their timing estimates, and (b) can modify their response strategies for artificial timing patterns.

1aPPb5. An analysis the actual sound and Foley sound at stepping on dead leaves. Ahn Iksoo, Myungjin Bae, and Seonggeon Bae (TeleCommun. & Information, soolsil Univ., 369 sangdo-ro, Dongjak-gu, Seoul 156-743, South Korea, aisbestman@naver.com)

The purpose of this research is to prove similarity between imitation sound of stepping on dead leaves used in media and actual sound of stepping on dead leaves and based on the result applying it to sound contents. By comparing imitation and actual sound, this research proves that imitated sound and the tools used for this are useful as sound contents. Also, it concludes that imitation sound of stepping on dead leaves has big potential to be developed as sound contents for not only performance, exhibition, and experience contents but also for treating spiritual health of people.

1aPPb6. Temporal model of edge pitch effects. Peter Cariani, H. Steven Colburn (Hearing Res. Ctr., Boston Univ., 44 Cummington St., Boston, MA 02215, cariani@bu.edu), and William M. Hartmann (Phys. and Astronomy, Michigan State Univ., East Lansing, MI)

Musical pitches can be perceived in broadband noise stimuli near frequencies corresponding to parameter changes along the frequency dimension. For example, monaural edge pitches (MEPs) are produced by noise stimuli with sharp spectral edges. Binaural edge pitches (BEPs) are produced by dichotic noise with interaural-phase changes at frequency boundaries (e.g., 0-pi), and binaural-coherence edge pitches (BICEPs) arise from boundaries between frequency regions with different interaural coherence (such as changes from correlated to uncorrelated noises). Perceived pitches are shifted slightly in frequency away from spectral edges: MEPs are shifted into noise bands, BICEPs are shifted into the incoherent region, and BEPs are shifted bimodally. This presentation proposes a temporal model for these edge pitches based on population-wide all-order interspike interval distributions (summary autocorrelations, SACFs), computed using the Zilany-Bruce-Carney 2014 model of the auditory nerve. Pitches were estimated from mean SACF values among lags at all F0-subharmonics (0–30 ms). Binaural pitches were estimated from computations based on corresponding left and right PSTs after a binaural delay and cancellation process. Model predictions agreed well with pitch judgments in both monaural and binaural cases. [Work supported by NIDCD (R01 DC00100 and P30 DC004663) and by AFOSR FA9550-11-1-0101.]

1aPPb7. Relative weight of temporal envelopes across speech frequency regions for speech intelligibility in hearing-impaired listeners and cochlear implant users. Yingjiu Nie, Harley J. Wheeler, Alexandra B. Short, and Caleb W. Harrington (Commun. Sci. and Disord., James Madison Univ., 801 Carrier Dr. - MSC 4304, Harrisonburg, VA 22807, nieyx@jmu.edu)

The study was aimed to investigate, among three groups of listeners—normal-hearing, hearing-impaired, and cochlear implant users, the relative weight of temporal envelopes for speech intelligibility in each of the eight frequency regions ranging between 72 and 9200 Hz. Listeners were tested in quiet and in the presence of steady or amplitude modulated noise at two rates (4 and 16 Hz). An eight-band vocoder was implemented when testing the acoustic-hearing groups. Speech intelligibility of a given region/band was assessed by comparing scores in two conditions differing only by the presence or absence of the band of interest; the proportion of the derived score to the sum across the eight regions/bands was computed as the relative weight. Preliminary data showed the following: (1) in quiet, similar frequency-weighting pattern for all three groups with higher weight in the mid/mid-high frequency range; (2) in noise, for the normal-hearing group, different weighting patterns between steady noise and amplitude-modulated noise; for the other two groups, similar weighting patterns for all types of noise with comparable weight across bands # 2–7 and lower weight for the bands # 1 and 8. The contribution of each region/band to masking release will also be discussed.

1aPPb8. Analytic and divided listening in normal-hearing and hearing-impaired listeners measured in a nonspeech pattern identification task. Elin Roverud, Virginia Best, Christine R. Mason, and Gerald Kidd, Jr. (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, erover@bu.edu)

In multisource listening environments, it is important to be able to attend to a single sound source (analytic listening) while concurrently monitoring unattended sources for potentially useful information (divided listening). Previous studies have indicated that hearing-impaired (HI) listeners have more difficulty with analytic listening than do normal-hearing (NH) listeners. Although central factors (e.g., selective attention and memory) clearly play a role, the extent to which differences in peripheral factors (e.g., auditory filter characteristics) contribute to this effect is not clear. In this study, performance in a closed-set nonspeech tonal pattern identification task was measured in NH and HI listeners for patterns centered at 850 and 3500 Hz. The frequency spacing among the tones forming the patterns was adjusted to equate performance across listeners at each center frequency separately to control for peripheral frequency resolution. Patterns were then played at both frequencies concurrently. Listeners were instructed to attend to either the low or high frequency and identify the pattern. In a second condition, patterns were randomly presented at one frequency with a foil at the other frequency, requiring the listener to monitor both frequencies to identify the pattern. Preliminary findings suggest that peripheral and central factors contributed to performance. [Support: NIH-NIDCD.]

1aPPb9. Voice emotion recognition and production by individuals with normal hearing and with cochlear implants. Monita Chatterjee, Aditya M. Kulkarni, Julie A. Christensen (Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, monita.chatterjee@boystown.org), Mickael L. Deroche, and Charles J. Limb (Otolaryngol., Johns Hopkins Univ. School of Medicine, Baltimore, MD)

Children and adults with normal hearing (NH) as well as those who use cochlear implants (CIs) were tested on a voice emotion recognition task. The materials were child-directed sentences, and acoustic analyses showed that features such as voice pitch range were exaggerated relative to earlier reports in the literature with adult-directed materials. The NH participants achieved ceiling-level performance in the task, while the children and adults with CIs had lower, and highly variable, scores. In a parallel study, we have also collected complex pitch discrimination thresholds in many of these participants. In a new preliminary study, we are analyzing the acoustic features of voice emotion production by these populations. The task involves reading simple sentences in a happy and a sad way. In this presentation, we will report on relationships between the perceptual data on voice emotion recognition and complex pitch discrimination by child and adult NH and CI participants. In addition, we will report on our initial acoustic analyses of voice emotion production by participants with NH and those with CIs.

1aPPb10. Spectral resolution and speech recognition in noise for children with hearing loss. Ryan W. McCreery (Audiol., Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, ryan.mccreery@boystown.org), Jenna M. Browning (Univ. of North Carolina, Chapel Hill, NC), Benjamin Kirby, Meredith Spratford, and Marc Brennan (Audiol., Boys Town National Res. Hospital, Omaha, NE)

Better spectral resolution has been associated with higher speech recognition in noise in adults with hearing loss who use hearing aids and adults with cochlear implants. However, the role of signal audibility and age on this relationship has not been reported. The goal of this study was to evaluate the effect of aided audibility and spectral resolution on speech recognition in noise for a group of children with sensorineural hearing loss and a group of adults with hearing loss. Higher age, better aided audibility, and the ability to detect more ripples per octave in a spectral ripple discrimination task were associated with better sentence recognition in noise for children with hearing loss.

1aPPb11. Application of the WKB method for an active cochlear model. Amir Nankali (Mech. Eng., Univ. of Michigan, 1092 Louise St., Apt. 14, Ypsilanti, MI 48197, nankali@umich.edu)

The Wentzel-Kramers-Brillouin (WKB) method has been used to approximate the solution for systems with slowly varying properties. The analytic method is computationally efficient and provides insights into the physical problem by decomposing the solution into a dominant wavenumber and the associated amplitude [e.g., Steele and Taber (1978)]. Because of its computational efficiency, this method provides a convenient means to estimate parameters in a complicated cochlear model, through variation and optimization. In turn, these parameters can be used in a more complete approximation technique (like a finite element or boundary element method) where parameter estimation can be more cumbersome. In this paper, we extend the WKB approximation to an active cochlear model, including the micromechanics of the organ of Corti (OoC) and electromotility. The model involves the OoC structural elements, basilar membrane (BM), and tectorial membrane (TM), coupled to the electrical potentials in the cochlear ducts and fluid pressure. Model predictions are compared to numerical approximations using finite elements. [Work supported by NIH-NIDCD R01-04084 and NIH NIDCD-T32-000011.]

1aPPb12. Priming gestures with associated sounds. Guillaume Lemaitre (Dept. of Psych., Carnegie Mellon Univ., IRCAM, 1 Pl. stravinsky, Paris 75004, France, GuillaumeJLemaitre@gmail.com), Nicole Navolio, and Laurie M. Heller (Dept. of Psych., Carnegie Mellon Univ., Pittsburgh, PA)

Hommel (1996) established that sounds artificially associated with key presses could prime key presses. The goal of the current study was to explore the nature of the link between auditory perception and manual actions by comparing the priming of manual actions by naturally and artificially associated sounds. We report three experiments. The procedure in each experiment consisted of cueing participants to perform manual gestures. Executing the gestures produced feedback sounds; those sounds were also used as primes before the response cues. Experiment One replicated Hommel's procedure: participants lifted keys that triggered artificial sounds. Experiment Two used sounds naturally produced by tapping or scraping wooden dowels on a custom interface. Experiment Three replicated Experiment Two with the sounds of the dowels muffled. The priming effect was observed on reaction times in Experiments One and Two but not in Experiment Three. These results show that long-term associations between sounds and gestures created in memory by repeated experience throughout life are not sufficient to prime the gestures. Instead they suggest that auditory-motor priming may be mediated by associations between sounds and gestures that are created on-line by the experiment via associative learning, and which are short-lived and can be readily reconfigured.

1aPPb13. Maximum equivalent pressure output and maximum stable gain of a light-activated contact hearing device that mechanically stimulates the umbo: Temporal bone measurements. Sunil Puria (EarLens Corp., 496 Lomita Mall, Stanford, CA 94305, puria@stanford.edu), Rodney Perkins (EarLens Corp., Menlo Park, CA), and Peter Santa Maria (Otolaryngology-HNS, Stanford Univ., Stanford, CA)

The non-surgical contact hearing device (CHD) consists of a light-activated balanced-armature tympanic-membrane transducer (TMT) that drives the middle ear through contact with the umbo, and a behind-the-ear unit with a widely vented light-emitter Assembly inserted into the ear canal that encodes amplified sound into pulses of light that drive and power the TMT. In comparison to acoustic hearing aids, the CHD is designed to provide higher levels of maximum equivalent pressure output (MEPO) over a broader frequency range and a significantly higher maximum stable gain (MSG). No artificial middle-ear model yet exists for testing the CHD, so we instead measured it using fresh human cadaveric temporal bones. To calculate the MEPO and MSG, we measured the pressure close to the eardrum and stapes velocity for sound drive and light drive using the CHD. The baseline sound-driven measurements are consistent with previous reports in temporal bones. The average MEPO ($N = 4$) varies from 116 to 128 dB SPL in the 0.7 to 10 kHz range, with the peak occurring at 7.6 kHz. From 0.1–0.7 kHz, it varies from 83 to 121 dB SPL. For the average MSG, a minimum of

about 10 dB occurs in the 1–4 kHz range, above which it rises as high as 42 dB at 7.6 kHz. From 0.2 to 1 kHz, the MSG decreases linearly from about 40 dB to 10 dB. The measurements of MEPO and MSG are compared with predictions from circuit-model calculations. The CHD may offer a way of providing broad-spectrum amplification appropriate to treat listeners with mild-to-severe hearing impairment.

1aPPb14. Evaluation of a navigational application using auditory feedback to avoid veering for blind users on Android platform. György Wersényi (Dept. of Telecommunications, Széchenyi István Univ., Egyetem tér 1, Györ 9026, Hungary, wersenyi@sze.hu)

Assistive technologies incorporate various applications and devices to assist handicapped people. This includes areas such as user interface design, sound design, mapping of visual and spatial information, sonification, etc. Visually impaired people also use state-of-the-art solutions especially for safe and independent navigation with the help of electronic travel aids (ETAs). Smartphones and tablets offer new possibilities in creating applications using the in-built sensors and open-source development platforms such as Android. This paper presents a short overview of recent technical approaches to assist users with visual impairment focusing on mobile applications for the Android platform. Evaluation of a navigational assistant using the magnetic sensor and tactile/auditory feedback to avoid veering is presented. Users can use the application after a short training time and keep straight walking path in outdoor and indoor tests using different sounds and vibration of the smartphone.

1aPPb15. Effect of some basic and premium hearing aid technologies on non-speech sound acceptability. Jingjing Xu, Jani A. Johnson, and Robyn M. Cox (School of Commun. Sci. and Disord., Univ. of Memphis, 807 Jefferson Ave., Memphis, TN 38105, jxu3@memphis.edu)

Acceptability of everyday non-speech sounds is closely related to hearing aid (HA) satisfaction. Acceptability is determined by a listener's overall impression of a sound when its different aspects, such as loudness, naturalness, and clarity, are considered. Various HA features, especially digital noise reduction (DNR), are designed to improve acceptability. Compared to basic HAs, premium HAs have more advanced DNR functions, as well as other unique features that are not included in basic HAs. Manufacturers often claim that everyday non-speech sounds are more acceptable when listening with premium HAs relative to basic HAs. However, there is minimal evidence to support this claim. This study evaluated acceptability of non-speech sounds in laboratory and real-world settings when using exemplars of basic and premium HAs. Forty-five older adults with mild-to-moderate hearing loss were bilaterally fitted with four pairs of BTEs (two basic and two premium) from two major manufacturers. Outcomes were obtained for each pair after a four-week field trial. Laboratory data were acceptability ratings of 21 real-time produced everyday sounds with different durations and intensities. Self-report data were rating scores from three questionnaires. No evidence was found in this study to show that premium HAs yield greater acceptability than basic. (Work supported by NIDCD.)

1aPPb16. Effect of spectral amplitude elevation for vowel identification in simulated combined electric-and-acoustic hearing. Fei Chen (Dept. of Elec. and Electron. Eng., South Univ. of Sci. and Technol. of China, Shenzhen, Guangdong 518055, China, fchen@sustc.edu.cn)

Though the benefits of combined electric-and-acoustic stimulation (EAS) have been widely reported, its underlying mechanisms responsible are not well understood yet. This study assessed the effect of balanced spectral amplitudes between acoustic and electric portions in EAS to vowel identification via two experiments. Four vowels (i.e., /iy/, /eh/, /oo/ and /ah/) were synthesized using pure-tone harmonic components. In Experiment 1, the spectral amplitudes of natural vowels were elevated for either acoustic (below 600 Hz) or electric portions (above 600 Hz) from 5 dB to 30 dB. Vowel identification scores were collected from eight normal-hearing listeners. In Experiment 2, the synthesized vowel stimuli were processed by the signal processing condition simulating the electric-and-acoustic stimulation. The results of identifying synthesized vowels and synthesized EAS-processed vowels in the two experiments showed the similar patterns of declined identification rates as a function of (acoustic or electric) spectral amplitude elevation. Furthermore, the specific loudness pattern computed from Moore *et al.*'s model was used to predict the vowel identification score, yielding a good prediction performance (i.e., $r = 0.9$). The results in the present work suggested the importance to maintain the balance between the spectral amplitudes of acoustic and electric portions for vowel identification in EAS hearing.

1aPPb17. Speech experience and cue trading in budgerigars. Mary M. Flaherty (Psych., SUNY Buffalo, 392 Park Hall, North Campus, Buffalo, NY 14260, maryflah@buffalo.edu), James R. Sawusch, and Micheal L. Dent (Psych., SUNY Buffalo, Amherst, NY)

The current project investigates how experience with human speech can influence speech perception in budgerigars. Budgerigars are vocal mimics and speech exposure can be tightly controlled in a laboratory setting. The data collected include behavioral responses from 30 budgerigars, tested using a cue-trading paradigm with synthetic speech stimuli. Prior to testing, the birds were divided into three exposure groups: Passive speech exposure (regular exposure to human speech), no speech exposure (completely isolated), and speech-trained (using the Model-Rival Method). After the exposure period, all budgerigars were tested using operant conditioning procedures. Birds were trained to peck keys in response to hearing different synthetic speech sounds that began with either "d" or "t." Sounds varied in VOT and in the frequency of the first formant. Once training performance reached 80% on the series endpoints, budgerigars were presented with the entire series, including ambiguous sounds. The responses on these trials were used to determine which speech cues the birds use, if cue trading behavior was present, and whether speech exposure had an influence on perception. Preliminary data suggest experience with speech sounds is not necessary for cue trading by budgerigars.

1aPPb18. Can you say that again? Aging and repetition effects in different types of maskers. Karen S. Helfer, Richard L. Freyman, Angela Costanzi, Sarah Laakso, and Gabrielle Merchant (Commun. Disord., Univ. of Massachusetts Amherst, 358 N. Pleasant St., Amherst, MA 01002, khelfer@comdis.umass.edu)

The most common strategy used by listeners when they cannot understand an utterance is a request for repetition. Repetition may improve perception in a number of ways, including priming the target message via sharpening the representation of aspects of the signal centrally, buying additional time for the listener to process the message, and enhancing the coding of information into memory. In the present study, the benefit of hearing an utterance a second time was measured in three different types of maskers (competing speech, single-channel envelope modulated noise, and steady-state speech shaped noise) using two different tasks: immediate recognition of the target sentence and memory for words presented during a previous block of trials. For the latter task, participants were presented with isolated words and were asked if those words appeared in the previous set of sentences. Participants (younger, middle-aged, and older adults) also completed a battery of cognitive tasks that included measures of processing speed, attentional control, and working memory. This presentation will describe results of analyses comparing the benefit of repetition for immediate recall and memory among the listener groups in the presence of different types of maskers. (Work supported by NIDCD R01 DC012057.)

1aPPb19. Listener consistency in identifying speech mixed with particular "bubble" noise instances. Michael I. Mandel, Sarah E. Yoho, and Eric W. Healy (Speech and Hearing Sci., The Ohio State Univ., 395 Dreese Labs, 2015 Neil Ave., Columbus, OH 43210, mandelm@cse.ohio-state.edu)

Previous work has shown that the intelligibility of mixtures of the same exact speech token with different instances of "bubble" noise is highly dependent on the configuration of the random time-frequency glimpses it provides. In the current study, the consistency of these judgments was measured for such mixtures involving six consonants, /t/, /d/, /f/, /v/, /t͡ʃ/, /d͡ʒ/, in an /a/-consonant-/a/ context. Intra-subject consistency on ten repetitions each of 60 mixtures was found to be high, as was inter-subject consistency between five normal-hearing subjects on 1200 mixtures. In addition, the noise level at a small subset of time-frequency points was significantly correlated with the overall intelligibility of the mixtures of a given token, suggesting that these regions contribute more strongly to correct identification than others. The current study finds that these regions are quite consistent across the five subjects. For example, for the token /ada/, these points surround the stop burst and the resolved harmonics following the onset of voicing. These results show the promise of the "bubbles" methodology for identifying time-frequency regions of individual utterances that contribute most to their intelligibility, permitting future study of such regions for different types of speech and listeners. [Work supported by NIH.]

1aPPb20. The effect of amplitude envelope on spatial ventriloquism. Dominique Beauregard Cazabon (Psych., Neurosci. & Behaviour, McMaster Univ., 424 Togo Salmon Hall, 1280 Main St. West, Hamilton, Ontario L8S 4M2, Canada, beaureda@mcmaster.ca) and Michael Schutz (School of the Arts, McMaster Univ., Hamilton, Ontario, Canada)

Spatial ventriloquism occurs when a visual event and an auditory event happen simultaneously; location judgments for the sound source become biased toward the location of the visual event. Evidence suggests the brain attributes greater weight to spatial information provided by the visual stimulus because the visual system offers better spatial acuity; when the visual stimulus is deteriorated, the visual bias is reduced. Thus, the brain performs optimal bimodal integration: greater weight is given to the modality which provides more information. The present study aims to determine whether the amplitude envelope of sounds provides spatial localization information to the perceptual system. We used a psychophysical staircase procedure to measure spatial ventriloquism experienced by participants for sounds with percussive, flat, and time-reversed percussive envelopes. We hypothesize that percussive and reverse-percussive sounds provide more information and thus better sound localization acuity than flat sounds, which would result in smaller degrees of spatial ventriloquism for the former than the latter. The results yield insight into the brain's use of auditory cues in audio-visual integration.

1aPPb21. Role of high-frequency component in source segregation in multi-talkers condition. Jing Mi and H. Steven Colburn (Biomedical Eng., Boston Univ., 44 Cummington Mall, Rm. 427, Boston, MA 02215, jingmi@bu.edu)

This study explores the role of high-frequency components in generating spatial release from masking (SRM) in multiple-talker conditions. From a frequency-analysis perspective, one can fully separate multiple speech signals when their spectrograms do not overlap. The sparsity of speech can be related to spectral overlap and is explored here in the context of SRM. The interaural coherence coefficient (IACC), which is defined as the value of the cross-correlation function at the expected interaural time difference (ITD) location, can be a good indicator of sparsity for binaural signals. The sparser the stimuli in the frequency domain, the less overlap in spectrograms, which leads to higher short-term IACCs. In this study, a sound-segregation system based on binaural cues was used to separate multiple sound signals in a mixture. Then, the short-term IACC was calculated for each time-frequency slice in the segregated sound. It was shown that average IACCs for multi-speech conditions are higher than for multi-white-noise conditions, especially in high-frequency regions. Based on this result, it was hypothesized that high-frequency components, which are sparser than low-frequency components in speech, are critical for SRM in multiple-talker conditions. Psychoacoustic evidence was found to support this hypothesis. [Work supported by NIH/NIDCD: Grant R01 DC00100.]

1aPPb22. The acoustic survey of intonation in Autism Spectrum Disorder. Zahra Azizi, (Linguist, Ferdowsi Univ. of Mashad, Modarres Boulevard – Dasht-e Chenar St. – Alley Number 7 – No 260, Shiraz 7158616976, Iran, zahra.azizima84@gmail.com)

The speech pattern of children with autism spectrum disorder (ASD) demonstrates a number of abnormal features, the study which can shed light on the difficulties the autistic children encounter in making meaning through speech. With regard to this stated purpose, in this article, I have scientifically examined the pattern of speech in a set of indicative and interrogative sentences as uttered by a group of autistic children and compared and contrasted the results with the pattern of speech in the same set of sentences as uttered by a group of typical development (TD) children. The specific aim of this study was to assess the intonation pattern, mean of pitch, amplitude, duration, intensity, and tilt in the ASD as compared to the TD. The data collected showed that while the amplitude in interrogative sentences, duration, intensity, mean of pitch, and tilt in ASD and TD were almost similar, the intonation pattern, the mean of pitch (with regard to severity of autism), and the amplitude in indicative sentences proved significantly different. To be more precise, with respect to these three latter features, the autistic children demonstrated monotony, an impairment that made them have difficulty making meaningful indicative and interrogative sentences.

1aPPb23. Envelope-power based prediction of auditory masking and speech intelligibility. Thomas Biberger and Stephan D. Ewert (Medizinische Physik and Cluster of Excellence Hearing4all, Carl-von-Ossietzky-Straße 9-11, Oldenburg, Lower Saxony 26135, Germany, thomas.biberger@uni-oldenburg.de)

In natural surroundings, people are often part of challenging acoustical scenarios (noise) interferers and reverberation. Amplitude modulations (AM) are fundamental acoustic features used to extract information in such situations. It has been shown that the envelope power spectrum model [EPSM; Ewert and Dau, J. Acoust. Soc. Am. **108**, 1181 (2000)] can account for psychoacoustic AM detection and masking data by considering the long-term envelope signal-to-noise-ratios (SNR) at the output of modulation filters. Recently, this concept was extended to consider short-term envelope SNRs to predict speech intelligibility in fluctuating maskers and in reverberation [mr-sEPSM; Jørgensen *et al*., J. Acoust. Soc. Am. **134**, 1475 (2013)]. In this study, the EPSM was further extended to consider envelope power SNRs (AM cues) on various time scales and "classic" power SNRs (intensity cues). The goal was to develop an auditory model for jointly predicting psychoacoustic masking and speech intelligibility. The model was demonstrated to account for a broad variety of psychoacoustic effects as AM detection, AM discrimination, AM masking, spectral masking, forward masking, intensity just-noticeable-differences (JNDs) and hearing threshold. Speech intelligibility was predicted comparable to Jørgensen *et al*.. Implications for future modeling including applications to audio and speech quality assessment are discussed.

1aPPb24. Individual differences in auditory brainstem response wave-V latency in forward masking: A measure of auditory neuropathy? Golbarg Mehraei (Health Sci. and Technol., Massachusetts Inst. of Technol., 23 Elm St. #5, Cambridge, MA 02139, gmehraei@gmail.com), Andreu P. Gallardo, Bastian Epp (Elec. Eng., Denmark Tech. Univ., Kgs. Lyngby, Denmark), Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA), and Torsten Dau (Elec. Eng., Denmark Tech. Univ., Kgs. Lyngby, Denmark)

A recent animal study suggests that noise exposure causes a preferential loss of low-spontaneous rate (low-SR) auditory nerve fibers (ANFs). This loss may leave detection thresholds normal yet degrade temporal encoding of supra-threshold sounds. Differences in the rate of recovery from forward masking in ANFs with different spontaneous rates may allow one to assess the state of different ANF populations. To test this, we measured auditory brainstem response (ABR) in a forward masking paradigm and evaluated wave-V latency changes with increasing masker-to-probe intervals (MPI). We expected that (1) loss of ANFs increases wave-V latency and forward masking thresholds, and (2) a preferential loss of low-SR fibers results in faster recovery time of wave-V latency. To test our hypotheses, we presented listeners with a broadband noise masker at two levels followed by a chirp probe at various MPIs. Initial results show that normal hearing threshold (NHT) listeners with delayed wave-V latency exhibit higher behavioral detection thresholds. Additionally, the listeners with the poorest behavioral thresholds had the fastest threshold recovery as a function of MPI. These results are consistent with the hypothesis that a preferential loss of low-SR fibers explains differences in NHT listeners.

1aPPb25. Masked speech recognition in a single-talker or a single-talker-plus-noise masker in school-age children and adults. Heather Porter (Otolaryngology/Head and Neck Surgery, The Univ. of North Carolina at Chapel Hill, 170 Manning Dr., CB 7070, Chapel Hill, NC 27599-7070, heather_porter@med.unc.edu), Lori J. Leibold (Allied Health, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC), and Emily Buss (Otolaryngology/Head and Neck Surgery, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

Previous work has shown that masked speech perception is poorer in younger children than older children and adults, particularly when the masker is composed of speech. The present study tested the hypothesis that young children's poor performance in speech maskers is due to reduced ability to benefit from transient improvements in the target-to-masker ratio (TMR). Listeners were normal-hearing school-age children (5–15 years) and adults. The target words and masker speech were produced by two different female talkers. Target stimuli were disyllabic words, and the masker was either: (1) speech-only or (2) speech-plus-noise. Both maskers included a continuous 60-dB-SPL stream of speech. The speech-plus-noise masker also included a continuous 50-dB-SPL speech-shaped noise. A four-alternative forced choice task was used, and target level was adapted to estimate threshold. Thresholds were higher for younger children than older children or adults for both masker conditions. While the inclusion of speech-shaped noise increased thresholds for all listeners, it had a smaller detrimental effect on younger children than on older children and adults. This result is consistent with the idea that greater susceptibility to speech-on-speech masking in younger children is related to a reduced ability to benefit from transient improvements in TMR.

1aPPb26. Dual-carrier vocoder: Effect on streaming in a cochlear implant simulation. Frederic Apoux, Brittney Carter, and Eric W. Healy (The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, fred.apoux@gmail.com)

Recently, it has been suggested that temporal fine structure (TFS) cues play a critical role in streaming while providing little speech information. To take advantage of this role of TFS, a coding strategy for cochlear implants (CIs) has been developed involving two independent carriers, one for the target and one for the background. The so-called dual-carrier coding can help to provide awareness of the background sounds to CI users while improving speech intelligibility. Two experiments evaluated the ability of normal-hearing (NH) listeners to process a target sentence and a "competing background" sentence and their ability to switch from one sentence to the other when presented with dual-carrier stimuli. The first experiment showed that NH listeners can deliberately focus their attention and successfully process the target or the background sentence. It also showed that this capability is not greatly affected by the addition of a 500 ms sentence fringe at the beginning of one of the sentences. Similarly, the second experiment showed that NH listeners could understand both sentences when presented with the same stimulus more than once. This last result confirms, if needed, that the target and background signals are both intelligible in dual-carrier stimuli. [Work supported by NIH.]

1aPPb27. An algorithm that generalizes to novel noise segments to improve speech intelligibility for hearing-impaired listeners. Eric W. Healy, Sarah E. Yoho (Speech & Hearing Sci., The Ohio State Univ., Pressey Hall Rm. 110, 1070 Carmack Rd., Columbus, OH 43210, healy.66@osu.edu), Jitong Chen, Yuxuan Wang, and DeLiang Wang (Comput. Sci. and Eng., The Ohio State Univ., Columbus, OH)

Machine-learning algorithms to extract speech from background noise hold considerable promise for alleviating limitations associated with hearing impairment. One of the most important considerations for implementing these algorithms into devices such as hearing aids and cochlear implants involves their ability to generalize to conditions not employed during the training stage. A major challenge involves the generalization to novel noise segments. In the current study, sentences were extracted from multi-talker babble and from cafeteria noise using an algorithm that estimates the ideal ratio mask by employing deep neural networks. Importantly, the algorithm was trained on segments of noise and tested using entirely different segments of the same nonstationary noise type. The training set was expanded through a noise-perturbation technique. Substantial benefit was observed for hearing-impaired listeners in both noise types, despite the use of unseen noise segments during the operational stage. Interestingly, normal-hearing listeners displayed benefit in babble but not in cafeteria noise. These results highlight the importance of evaluating these algorithms not only in human subjects, but in members of the actual target population. [Work supported by NIH.]

1aPPb28. Does providing more processing time improve speech intelligibility in hearing-impaired listeners? Virginia Best, Christine R. Mason, Jayaganesh Swaminathan, Elin Roverud, and Gerald Kidd (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, ginbest@bu.edu)

There is ample evidence in the literature that hearing loss increases the time taken to process speech (e.g., increased response times for word discrimination, sentence identification, and passage comprehension). This has led to the assumption that providing hearing-impaired listeners with more time to process speech would be beneficial. For sentence identification, a number of studies have examined the effects of adding silent gaps, or using time-expansion, but the results have been mixed. This may be because the distortion of natural sentence structure counteracted any benefits, or because the tasks used were not sufficiently demanding. In the current study, we re-examined this issue in young hearing-impaired listeners. By using sequences of words spoken in isolation, we were able to insert pauses between words without disrupting natural word boundaries. Furthermore, we used speech tasks involving interference in order to reveal effects that might only be apparent when the processing load is high. Although we found a statistically significant benefit of providing silent gaps between words, it was modest and was only present for some listeners in certain conditions. Overall, there was little evidence that the benefit was larger for hearing-impaired listeners or under more challenging conditions. [Work supported by NIH/NIDCD.]

1aPPb29. Speaking while listening: Broadbent revisited. Nandini Iyer, Eric Thompson, Brian Simpson, and Griffin Romigh (Air Force Res. Lab., 2610 Seventh St., BLDG 441, Wpafb, OH 45433, Nandini.Iyer.2@us.af.mil)

Air Force operators are often required to listen to several channels of ongoing radio communications while simultaneously responding to some or all of these messages. Broadbent (1952) showed that performance in such speaking-while-listening tasks typically resulted in a significant degradation. However, it was not clear if this decrement in performance was due to a competition of the resources required for simultaneous message comprehension and response formulation, or some other factor influencing the representation (such as poor signal quality). A replication of Broadbent's study was undertaken to measure the ability of subjects to perform in speaking-while-listening tasks. Listeners responded to a series of yes-no queries posed by a recorded talker regarding the presence/location of items on a visual display. Responses were scored based on a subject's ability to (1) respond to their assigned call-sign, (2) use the correct querier call sign in their response, and (3) provide a response to the question about the query item. Two variables were manipulated: rate of incoming messages and fidelity of the recordings (additive noise or radio communications). Results indicate that changing the rate of incoming messages had the largest impact on accuracy of responses, while the fidelity of the message had a relatively minor impact on performance, suggesting that limitations in these tasks may be largely due to an interference with response formulation.

1aPPb30. The role of age and executive function in auditory category learning. Rachel Reetzke (Commun. Sci. and Disord., Univ. of Texas at Austin, 2504A Whitis Ave., Austin, TX 78712, rreetzke@gmail.com), Todd Maddox (Psych., Univ. of Texas at Austin, Austin, TX), and Bharath Chandrasekaran (Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX)

Auditory categorization is a natural and adaptive process that allows for the organization of high-dimensional, continuous acoustic information into discrete representations. The aim of this study is two-fold: (a) to examine the developmental trajectory of rule-based auditory category learning from childhood through early adulthood, and (b) to examine the extent to which individual differences in rule-based category learning relates to individual differences in executive function. Sixty participants with normal-hearing, 20 children (age

range, 7–12), 20 adolescents (age range, 13–19), and 20 young adults (age range, 20–23), learned to categorize novel spectrotemporally modulated sounds using trial-by-trial feedback. The experimental design included six blocks of 100 stimuli for a total of 600 trials. Results revealed that auditory categorization accuracy improved with age, with young adults outperforming children and adolescents. Computational modeling analyses indicated that the use of the task-optimal strategy (i.e., multidimensional learning strategy) improved with age. Further, correlational analyses revealed that the use of optimal multidimensional learning strategies was strongly correlated with individual differences in executive function, even when age was partialed out. The current findings demonstrate the protracted development of rule-based auditory categorization. The results further suggest that executive function strongly relates to successful strategy use during auditory category learning.

1aPPb31. Effect of age and hearing loss on intelligibility enhancement in a loudspeaker-based simulated reverberant environment. Nirmal Kumar Srinivasan (National Ctr. for Rehabilitative Auditory Res., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, nirmal.srinivasan@va.gov), Pavel Zahorik (Univ. of Louisville, Louisville, KY), Kasey M. Jakien, Samuel Gordon, Sean D. Kampel, Megan Stansell, and Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland, OR)

The ability of the human auditory system to adapt to changes in the modulation spectrum of a reverberant signal has been documented (Zahorik *et al.*, 2012). Previously, results were presented examining the effects of age and hearing loss on adaptation to reverberation in an anechoic chamber with spatially separated speech with monaural distortions only. The results indicated that older listeners obtained decreased benefit from prior exposure to listening environments. However, in a realistic reverberant environment, both monaural distortions and spatial cues are present and it is important to analyze the effects of reverberation in the presence of spatial cues. In this experiment, a more spatially realistic sound field using 24 speakers separated by 15° in azimuth was created in an anechoic chamber by calculating the directions, attenuations, and delays of the reflections from a sound source. An impulse response for each loudspeaker that has the appropriate delays and attenuations for all reflections in its spatial vicinity was created. Speech was convolved with individual speakers' impulse responses for different spatial separations between target and maskers before presenting to the listeners. Effects of age and hearing loss on intelligibility enhancement due to prior exposure to a realistic listening environment will be discussed. [Work supported by NIH R01 DC011828.]

MONDAY MORNING, 18 MAY 2015 KINGS 4, 9:00 A.M. TO 12:00 NOON

Session 1aSC

Speech Communication and Psychological and Physiological Acoustics: Listening Effort I

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

Christian Fullgrabe, Cochair
Institute of Hearing Research, Medical Research Council, Science Road, Nottingham NG7 2RD, United Kingdom

Chair's Introduction—9:00

Invited Papers

9:05

1aSC1. Research on listening effort: History and methods, theory, and practice. Alexander L. Francis (Purdue Univ., Speech, Lang. and Hearing Sci., Lyles-Porter Hall, 715 Clinic Dr., West Lafayette, IN 47907, francisa@purdue.edu) and Christian Füllgrabe (MRC Inst. of Hearing Res., Nottingham, United Kingdom)

The notion of listening effort has enjoyed a resurgence in recent years, with new work appearing in both theoretically and clinically oriented venues, and researchers pursuing this topic with an increasingly inventive suite of methods. Listening effort is often associated with perceptual and/or cognitive processing demands for understanding speech, terms that are often (but not necessarily) identified with mechanisms such as attention and working memory. Much of the recent research involving listening effort has arisen in the context of studies of people with hearing impairment and of people listening in adverse or sub-optimal conditions, thus highlighting a particular need to distinguish between measures of effort and more traditional measures of listening performance such as intelligibility. However, there is considerable terminological variety within the field, and significant debate concerning both the theoretical characterization and the clinical implications of all of these concepts. The goal of this talk is to provide a broad introduction to current research on listening effort, situating it within the context of a longer history of psychological studies of human performance and task demands and surveying some of the more common behavioral and psychophysiological methods used to assess listening effort for research and clinical purposes.

9:25

1aSC2. Listening effort and the division of auditory processing resources. Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov)

When a listener is asked to simultaneously perform two different tasks, the results can reflect a wide range of factors. This presentation will describe data showing that performance reflects not only the division of a single resource shared between the tasks, but also the structure of the tasks themselves. Thus, if two tasks share structural resources (such as the same auditory pathway or the same memory representations) there may be interference even if there is still spare capacity at the level of other cognitive resources. Similarly, reduced effort may reflect independent structural constraints rather than increased capacity. For example, if the structure of the tasks allows the listener to rely upon a memory representation for one or both tasks, then a task in which the stimuli are presented simultaneously can be processed serially by relying on the memory trace to do the secondary task. The implications of this framework for studies examining listening effort will be discussed, along with ideas about ways to improve the design of experiments seeking to probe the capacity of a single shared resource.

9:45

1aSC3. When a competing talker is easy to ignore: The elusive nature of informational interference. Sven Mattys (Dept. of Psych., Univ. of York, York YO10 5DD, United Kingdom, sven.mattys@york.ac.uk)

A competing talker can impair speech processing through both energetic masking and informational, cognitive aspects of the competing utterance. We refer to the latter as informational interference. We hypothesized that handling informational interference depletes processing resources that could otherwise be allocated to recognizing the target speech. As a consequence, informational interference should be particularly pronounced for target sentences with high processing demands (syntactically complex sentences) than for sentences with low-processing demands (syntactically simple sentences). Using a speeded picture selection task, we assessed native and non-native listeners' understanding of subject-relative (simple) and object-relative (complex) sentences played against a competing talker vs. a matched energetic mask. While object-relative sentences were more difficult to process than subject-relative sentences, there was no effect of masker type, and this pattern was comparable for native and non-native listeners and across various SNRs. Thus, contrary to prior research, we found no evidence that a competing talker requires greater processing resources than energetic masking alone. Ongoing eye-tracking and pupillometric versions of this experiment will establish the nature of the discrepancy and the conditions under which informational interference is absent.

10:05

1aSC4. Downstream effects of accented speech on memory. Kristin Van Engen (Washington Univ. in St. Louis, One Brookings Dr., Campus Box 1125, Saint Louis, MO 63130-4899, kvanengen@wustl.edu)

When speech is acoustically challenging, listeners frequently engage additional cognitive processes to effectively perceive it. This increased listening effort can have downstream effects on behavior, impairing listeners' memory for what they have heard. To date, the listening effort literature has focused on processing speech in challenging listening situations that involve signal degradation due to noise and/or hearing impairment. A common theme in this literature is that acoustic degradation results in speech signals that deviate from listeners' stored phonological and lexical representations—a characteristic that degraded speech shares with speech produced in an unfamiliar accent. In this talk, we bring accented speech into the broader conversation about listening effort. In particular, we present preliminary results investigating the effects of intelligible foreign-accented speech on listeners' subsequent memory for words and sentences, even when these stimuli are intelligible. This work allows us to begin to incorporate listener effort associated with accented and degraded speech into a unified cognitive framework.

10:25–10:40 Break

10:40

1aSC5. The relationship between phonetic cue weighting and listening effort in listeners with cochlear implants. Matthew Winn, Jan Edwards, and Ruth Litovsky (Waisman Ctr., Univ. of Wisconsin-Madison, 1500 Highland Ave., Rm. 565, Madison, WI 53705, mwinn83@gmail.com)

In the study of speech perception, measures of cognitive load are especially useful because speech intelligibility performance results from a wide variety of auditory and cognitive processes that might demand different amounts of effort. The experience of elevated effort is an important part of hearing impairment and could be the target of audiological outcome measures. Listeners demonstrate different cue-weighting strategies when identifying phonemes, which is especially apparent for cochlear implant patients, who receive a highly degraded signal. This study explored whether changes in signal processing that affect phonetic cue weighting are also associated with changes in listening effort. Using both a conventional and an experimental speech processing strategy in alternating blocks, listeners completed tests of basic intelligibility, cue weighting for the /b/-/d/ contrast, and listening effort indicated by pupil dilation. Listeners for whom the experimental strategy promoted increased weighting of formant transition cues generally also showed reduced effort when using that strategy to listen to complete sentences, compared to listening with their basic strategy. Intelligibility scores generally did not show much change across both strategies. Results suggest that intelligibility measures can be complemented by other measures that tap into the mechanisms of perception and the effort demanded by the process.

11:00

1aSC6. Factors that increase processing demands when listening to speech. Ingrid Johnsrude (Commun. Sci. and Disord., Western Univ., Rm. 227 Brain and Mind Inst., Natural Sci. Ctr., Western University, London, Ontario N6A 5B7, Canada, ijohnsru@uwo.ca) and Jennifer M. Rodd (Psych. and Lang. Sci., Univ. College London, London, United Kingdom)

Listening effort is an increasingly important concept for applied hearing researchers, but it has not been well elaborated in the cognitive literature. We propose that listening effort is the product of two factors: the processing demands imposed by the listening situation, and the cognitive resources that an individual brings to bear, to compensate for demands. Whereas cognitive resources differ markedly among individuals, processing demands are generally constant in a given listening situation, at least for normal-hearing individuals, and fall into at least three different categories: (i) perceptual demands, (ii) linguistic demands, and (iii) concurrent task demands. Perceptual demands are increased when the speech is degraded, when concurrent interfering sounds are present, or when the listener is hearing impaired. Linguistic demands are increased, for example, when speech is semantically or syntactically complex, or when meaning is ambiguous. Finally, when a listener is performing another task (such as driving) while hearing speech, additional demands are present. A series of behavioral and neuroimaging experiments reveals the nature of these different demands, which appear to be largely independent but to interact in left inferior frontal cortex; fMRI activity in this area may serve as an objective, quantifiable, measure of listening effort. Interactions among different types of demand imply that they cannot be fully understood by studying them in isolation.

11:20

1aSC7. Recruitment of the speech motor system in adverse listening conditions. Claude Alain and Yi Du (Rotman Res. Inst., Baycrest Hospital, 3560 Bathurst St., Toronto, Ontario M6A 2E1, Canada, calain@research.baycrest.org)

Background noise is detrimental to speech comprehension. The decline-compensation hypothesis posits that deficits in sensory processing regions caused by background noise can be counteracted by compensatory recruitment of more general cognitive areas. We are exploring the role of the speech motor system as a compensatory mechanism during impoverished sensory representations in the auditory cortices. Prior studies using functional magnetic resonance imaging (fMRI) have revealed an increase in prefrontal activity when peripheral and central auditory systems cannot effectively process speech sounds. Our research using event-related fMRI revealed a negative correlation between brain activation and perceptual accuracy in speech motor regions (i.e., left ventral premotor cortex (PMv) and Broca's area), suggesting a compensatory recruitment of the motor system during speech perception in noise. Moreover, multi-voxel pattern analysis revealed effective phoneme categorization in the PMv and Broca's area, even in adverse listening conditions. This is in sharp contrast with phoneme discriminability in auditory cortices and left posterior superior temporal gyrus, which showed reliable phoneme classification only when the noise was extremely weak. Better discriminative activity in the speech motor system may compensate for the loss of specificity in the auditory system by forward sensorimotor mapping during adverse listening conditions.

11:40

1aSC8. Electroencephalographic estimates of listening effort for hearing aid fitting. Daniel J. Strauss, Corinna Bernarding (Systems Neurosci. & NeuroTechnol. Unit, Saarland Univ., Saarland University Hospital, NeuroCtr. Bldg. 90.5, SNN-Unit, Homburg/Saar 66421, Germany, daniel.strauss@uni-saarland.de), Ronny Hannemann (Siemens Audiologische Technik, Erlangen, Germany), and Farah I. Corona-Strauss (Systems Neurosci. & NeuroTechnol. Unit, Saarland Univ., Saarbruecken, Germany)

Objective estimates of listening effort could support hearing aid fitting procedures. For this reason, we have analyzed neural correlates of listening effort using electroencephalographic methods and quantitative neurofunctional modeling in recent years. In particular, we have developed a neurophysical corticothalamic feedback model for electroencephalographic listening effort correlates in event-related potentials (ERP), which allowed us to compare simulated and measured data. However, ERP paradigms suffer from their limited flexibility in hearing aid fitting as they are restricted to a certain class of stimulation protocols. More recently, we have shown that attention related listening effort correlates can also be extracted from the oscillatory electroencephalographic activity when using a circular analysis of the instantaneous phase organization of Hardy space projected versions of the neural signals. In this talk, we are going (a) to review these approaches, (b) present results that show that the oscillatory method allows for the extraction of listening effort correlates in real life fitting settings, and (c) discuss the potential of time-resolved listening effort profiles. We think that the latter will stimulate new ideas what objective methods can do and what subjective psychometric methods cannot do.

Session 1aUW

Underwater Acoustics: Acoustic Communications and Scattering

Shane Guan, Cochair
National Marine Fisheries Service, 1315 East-West Highway, SSMC-3, Suite 13700, Silver Spring, MD 20902

Sean Walstead, Cochair
ECE/SIO, UCSD, 9500 Gilman Drive, 0407, La Jolla, CA 92093-0407

Contributed Papers

8:30

1aUW1. Experimental results on underwater communication using vector transducers. Erjian Zhang, Ali Abdi (Elec. Comput. Eng., New Jersey Inst. of Technol., Newark, NJ 07102, ez7@njit.edu), and Chen Chen (Samsung Mobile Solution Lab, San Diego, CA)

Recent studies show that a vector transducer which excites signals in acoustic particle velocity channels can serve as a compact multichannel communication transmitter (C. Chen and A. Abdi, "Signal transmission using underwater acoustic vector transducers," *IEEE Trans. Signal Processing*, **61**, 3683–3698, 2013). In this paper, performance of frequency shift keying modulation is studied, both theoretically and experimentally, for data transmission via underwater particle velocity channels using a vector transducer. Multiple test environments are considered. Comparisons are also made with systems that utilize scalar transducers only. System and transducer design considerations, implementation details, and experimental results are presented in the paper as well. [Work supported in part by the National Science Foundation (NSF), Grants IIP-1340415 and CCF-0830190.]

8:45

1aUW2. Blind deconvolution of simple communication signals recorded in laboratory water tank. Jane Kim, Alex Douglass, Paul Choi, and David R. Dowling (Mech. Eng., Univ. of Michigan, 760 Peninsula Ct., Ann Arbor, MI 48105, janehjki@umich.edu)

Overlapping synthetic time reversal (OSTR) is a technique for estimating the original source signal from acoustic array recordings in an unknown, multipath, time-varying sound channel without using a channel probe or calibration pulse. The OSTR technique applies synthetic time reversal (STR) to short-duration overlapping time segments of the received signal and then stitches the sequences of signal estimates together to blindly recover the original long-duration signal. Previous shallow ocean simulations of OSTR with a 2 kHz center frequency were successful. This presentation describes experimental results for OSTR in the reverberant environment provided by a laboratory water tank with 1.07-m diameter and filled to 0.80-m depth. Here, a variable-duration 50 kHz center frequency communication signal modulated with binary phase shift keying and four carrier cycles per bit was broadcast from an omnidirectional source to either a 1-by-16 or 4×4 receiving array that was 30 to 50 cm away. The nominal multipath reverberation time was varied from 10 ms to 4 ms by adding absorption material to the walls of the tank. Experimental signal reconstruction results are promising and demodulation diagrams are shown for 8, 64, 512, and 4096-bit signals. [Sponsored by the ONR and NAVSEA.]

9:00

1aUW3. Inter-pulse noise field during an Arctic shallow-water seismic survey. Shane Guan (National Marine Fisheries Service, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20902, shane.guan@noaa.gov), Joseph F. Vignola, John A. Judge, DIego Turo (Dept. of Mech. Eng., Catholic Univ. of America, Washington, DC), and Teresa J. Ryan (Dept. of Eng., East Carolina Univ., Greenville, NC)

Marine seismic surveys using airgun arrays generate intense underwater acoustic pulses. Those pulses may cause hearing impairment and/or behavioral disturbances in marine mammals. Few studies have investigated the resulting multipath propagation and reverberation from these pulses. This research uses acoustic recordings collected in a shallow region of the Beaufort Sea during an open-water seismic survey to characterize the noise field between airgun pulses. The disturbances between pulses are collectively referred to as the inter-pulse sound. Two methods were used to quantify the inter-pulse sound: the incremental computation method and the Hilbert transform method. The incremental computation method calculates the root-mean-squared sound pressure level in various sub-intervals, and the Hilbert transform method is used to calculate instantaneous acoustic amplitudes. Analyses using both methods yield identical results, showing that the inter-pulse sound field exceeds ambient noise levels by as much as 9 dB during relatively quiet conditions. The results also indicate that inter-pulse noise levels are related to the source distance. These two methods can be used to quantify the impact of anthropogenic transient noises on the acoustic environment and to address acoustic masking in marine mammals.

9:15

1aUW4. Multichannel blind deconvolution of sound source of opportunity in ocean waveguide. Sung-Hoon Byun, Karim G. Sabra (School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332, byunsh@kriso.re.kr), Ning Tian, and Justin Romberg (School of Elec. and Comput. Eng., Georgia Inst. of Technol., Atlanta, GA)

Signals that travel through an ocean waveguide are typically distorted when they are received by a remote receiver because of interference and distortion arising from multiple propagation paths. This presentation investigates the applicability of a physics-based blind-deconvolution technique proposed by Sabra *et al.* (JASA **127**, EL42, 2010) to sound sources of opportunity such as randomly radiating ships. Using only the noisy recorded signals from an underwater receiver array, this blind deconvolution provides a means to estimate the Green's functions between the source of opportunity and the elements of the receiver array as well as the original signal radiated by the source of opportunity. These estimated Green's functions can then be used for acoustic characterization of the ocean waveguide parameters which is typically done using controlled sources only. We will discuss the performance of the proposed approach using both numerical simulations and at-sea data using discrete shipping events recorded on a vertical array as sources of opportunities.

9:30

1aUW5. High-frequency broadband seafloor backscatter in a sandy estuarine environment. Eric J. Bajor (Univ. Of New Hampshire, 74 Rolling Ridge Ln., Methuen, Massachusetts 01844, ebajor@ccom.unh.edu), Thomas C. Weber, Larry Ward, Yuri Rzhanov, and Han Hu (Univ. Of New Hampshire, Durham, NH)

Seafloor backscatter collected with high-frequency (>100 kHz) hydrographic echosounders has become an important aspect of seafloor characterization for benthic ecologists and other scientists. The mechanisms that control acoustic scattering at these high frequencies are not completely understood, although surficial roughness and the presence of large scatterers (e.g., shell hash) are likely contributors. To further our understanding of these mechanisms, broadband (100–250 kHz) acoustic measurements were taken at a grazing angle of 45 degrees in a shallow-water, sandy environment with a known presence of shell hash. Stereo imagery was collected simultaneously to quantify the roughness spectrum of the seafloor. Sediment samples were also collected on site of the experiment to quantitatively analyze the content of shell hash. The frequency dependence of the seafloor backscatter will be discussed in terms of its consistency, or lack thereof, with surficial roughness and shell-fragment scattering models.

9:45

1aUW6. Statistical characterization of biologic clutter from the shallow water TREX13 reverberation experiments. John R. Preston (ARL, Pennsylvania State Univ., P. O. Box 30, MS3510, ARL, Penn State Univ., State College, PA 16804, jrp7@arl.psu.edu)

A large experimental effort called TREX13 was conducted in April–May 2013 off Panama City, Florida. As part of this effort, reverberation and clutter measurements were taken in a fixed-fixed configuration in very shallow water (~20 m) over a 22 day period. Results have been presented previously characterizing reverberation, clutter and noise in the 1800–5000 Hz band. The received data are taken from the triplet sub-aperture of the Five Octave Research Array (FORA). The array was fixed 2 m off the sea floor and data were passed to a nearby moored ship (the R/V Sharp). An ITC 2015 source transducer was fixed 1.1 m off the seafloor nearby. Pulses comprised of gated CWs and LFMs were used in this study. Matched filtered plots of the reverberation and clutter along particular bearings are presented using the FORA triplet beamformer. There are clear indications of biologic scattering. Statistical characterization of the biologic components of the reverberation are presented using K-distribution based algorithms to note differences in the estimated shape parameter. Help from the Applied Physics Laboratory at the University of Washington was crucial to this effort. [Work supported by ONR code 322OA.]

10:00

1aUW7. Statistics of very high frequency sound scattered from wind-driven waves. Sean Walstead (ECE, UCSD, 9500 Gilman Dr., 0407, La Jolla, CA 92093-0407, swalstead@ucsd.edu) and Grant B. Deane (SIO, UCSD, La Jolla, CA)

The amplitude, Doppler spread, and temporal coherence of VHF scattering is important for the performance of high frequency sonars and underwater communications systems in operating scenarios where energy from the sea surface cannot be screened. For this talk, the amplitude and arrival time statistics of very high frequency (50 kHz–2000 kHz) sound scattered from a rough wind-driven wave surface are modeled. Saturation occurs in the normalized second moment of acoustic intensity when the surface correlation length exceeds a Fresnel zone length. Fluctuations in arrival time do not saturate and increase proportionally to the dominant surface wave component. Data collected at 300 kHz agree with the model results. Energy is first detected in the wind-driven surface wave field at wavelengths of 1.7 cm and 3.4 cm. The former wave corresponds to a gravity-capillary wave whose phase speed is at a minimum. The principal appearance of this type of wave has threshold implications for the range of acoustic transmission frequencies susceptible to saturation in amplitude statistics.

10:15

1aUW8. The effect of internal waves on the ambient noise vertical directionality in deep ocean. Mehdi Farrokhrooz and Kathleen E. Wage (George Mason Univ., 4450 Rivanna River Way PMB3740, Fairfax, VA 22030, mfarrokh@masonlive.gmu.edu)

Vertical array measurements of low-frequency ambient noise in the mid-latitudes indicate that distant shipping noise has a flat angular distribution concentrated around broadside, e.g., [Wales and Diachok, JASA, 1981]. Ships cannot directly excite the lowest modes (associated with broadside angles) since those modes are trapped near the sound channel axis. Dashen and Munk [JASA, 1984] considered three mechanisms that could transfer shipping noise into the low modes: (1) downslope conversion on the continental slope, (2) volume scattering due to internal waves, and (3) direct excitation of the low modes at high latitudes where the sound channel intersects the surface. Dashen and Munk conclude that the first mechanism is the most likely since the second requires impractically large propagation ranges and the third requires unrealistic shipping densities at high latitudes. A limitation of Dashen and Munk's internal wave analysis is that it does not consider the effect of bottom loss. This talk adapts the transport theory approach of Colosi and Morozov [JASA, 2009] to study the effect of internal waves on the vertical directionality of distant shipping noise in the presence of seafloor attenuation. Initial results indicate that internal waves may have a significant effect at shorter ranges than previously thought.

10:30–10:45 Break

10:45

1aUW9. Nonlinear interaction of monochromatic wave with noise. Desen Yang, Qing Liu, Shengguo Shi, Jie Shi, and Bo Hu (College of Underwater Acoust. Engineefing, Harbin Eng. Univ., No.145 Nantong St., Nangang District, Harbin City, Heilongjiang Province, Harbin, China, liuqing0104@126.com)

The nonlinear interaction of monochromatic wave with noise is studied theoretically, and that is based on the theory of O. V. Rudenko about interactions of intense noise waves and the theory of G. H. Du about nonlinear interaction of finite amplitude irregular waves. The effect on shifting of noise energy by the intensity and frequency of monochromatic wave is discussed during the propagation of the monochromatic wave and noise. Some numerical simulations about this process are conducted under different initial conditions to investigate the changes of their spectrum. The agreement between theoretical analysis and numerical simulation demonstrates that under some conditions the noise can be modulated and the noise spectrum becomes flatter and broader with the increasing distance.

11:00

1aUW10. Two-dimensional modeling of sound scattering with corrective smoothed particle method. Xu Li, Tao Zhang (Dept. of Naval Architecture and Ocean Eng., School of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan 430074, China, lixu199123@gmail.com), and YongOu Zhang (Dept. of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei, China)

Meshfree methods are widely used in solving acoustic problems, since a set of arbitrary distributed particles can easily represent systems with moving boundaries or a variety of media. Among all meshfree methods, the smoothed particle hydrodynamics (SPH) method is suitable in handling problems with large ranges of density and object separation. This paper aims at using the corrective smoothed particle method (CSPM) to simulate the two-dimensional underwater sound scattering in the time domain. First, a novel kind of acoustic boundary for the CSPM is built and tested with a plane wave model. Then, the CSPM code is used to solve sound scattering field of an infinite long cylinder, which has different computational parameters for the boundary condition. Finally, the distribution of sound scattering pressure obtained with theoretical solutions is used to validate the CSPM code, and the efficiency of the code is analyzed by comparing with the SPH results in the fields of computation time and accuracy. The suitable value of computational parameters of the boundary is also discussed.

11:15

1aUW11. Design optimization of low density parity check codes in multipath underwater acoustic channels. Shengxing Liu and Qiang Fu (Appl. Ocean Phys. and Eng., Xiamen Univ., Xiping Bldg. C3-223, Xiamen University, Xiamen 361102, China, liusx@xmu.edu.cn)

We propose an iterative decoding and equalizing scheme using low density parity check codes and decision feedback equalizer for underwater acoustic communication system. The scheme decodes and equalizes iteratively by linking a LDPC decoder with an adaptive DFE and feeding output soft information of the LDPC decoder to the DFE as priori information. We extend extrinsic information transfer (EXIT) charts to analyze the performance of LDPC codes in multipath underwater acoustic channels. Furthermore, we introduce an EXIT aided method of optimally designing the parameters of LDPC codes using modified differential evolution algorithm in multipath underwater acoustic channel. Design example and simulation are presented in two different underwater acoustic channels which are generated by BELLOP ray tracing model. Bit error rate of the proposed scheme is less than 10^{-5} as signal to noise rate (SNR) is greater than 8.5 dB in the channels. Compared the regular LDPC code with degree 3 and the Turbo code with feedback and feedforward generator polynomials $1+D+D^2+D^3$ and $1+D^2+D^3$, the optimized LDPC codes achieve gains of about 0.5 dB and 0.8 dB, respectively, at BER= 10^{-4} under the same code rate and block length.

11:30

1aUW12. Experimental results of adaptive multichannel decision feedback equalization in shallow channel. XueLi Sheng, LiNa Fan (Harbin Eng. University, No.145 Nantong St., Nangang District, Harbin, HeiLongjiang 150001, China, shengxueli@aliyun.com), Aijun Song, and Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Newark, DE)

A kind of underwater communication receiver combined with mean-squared error adaptive error adaptive algorithm and multichannel decision feedback equalization (AMDFE) is studied, which can address the problem of underwater communications over a time-varying multipath channel in the presence of worse intersymbol interference and waveform distortion. Its performances in different range and sound profile have been analyzed by simulated shallow water channels in the earlier paper. In this paper, the performances of AMDFE in different underwater channels with different parameters are mainly demonstrated by the experiments on the lake and sea. The anti-multipath ability and parameters selection regulation of AMDFE in both near and long range are proved by real data in shallow water.

11:45

1aUW13. Experimental demonstration of multiband underwater acoustic transmissions based on single carrier communication. Xiao Han, Jingwei Yin, Ge Yu, and Bing Liu (Harbin Eng. Univ., No. 145 Bldg., Nantong St., Nangang District, Harbin, Harbin 150001, China, hanxiao1322@hrbeu.edu.cn)

In a recent shallow water experiment, multiband transmissions were carried out using a vertical array with six sensors uniformly spaced every 0.5 m. The entire frequency band (2–8 kHz) was divided into two sub-bands, each of which is about 2.55 kHz in width. Time reversal processing is first used to compress severe multipath spread, following by a single channel decision feedback equalizer (DFE) to remove residual inter-symbol-interference (ISI). Using experimental data, this paper demonstrates that multiband transmissions are very beneficial to improve the throughput of underwater acoustic communications, achieving a data rate of 6k bits/s using QPSK modulation with almost error free performance.

Posters will be on display and contributors will be at their posters during the session break from 10.30 a.m to 10.45 a.m.

1aUW14. Experimental demonstration of under ice acoustic communication. Jingwei Yin (Harbin Eng. Univ., Rm. 301, Shuisheng Bldg., Harbin 150001, China, yinjingwei@hrbeu.edu.cn), Pengyu Du, Xin Wang, and Jiahao Guo (Harbin Eng. Univ., Harbin, Heilongjiang, China)

Under ice acoustic communication experimental was done in Songhua River, Harbin, China, in January 2015. Minus 20–30 degrees work environment brings a great challenge to the under ice experimental. A series of underwater acoustic communication tests including spread spectrum, OFDM, pattern time delay shift coding (PDS) and CDMA have been achieved. Experimental result shows that the under ice channel is relatively stable and the closer to the ice the simpler the channel structure is. All of the under ice acoustic communication tests achieve low bit error rate communication at 1 km range with different received depth. Under ice CDMA multiuser acoustic communication shows that as many as 12 users can be supported simultaneously with as few as five receivers in under ice channels, using the time reversal mirror combined with the differential correlation detectors.

1aUW15. Fading statistics variation and optimum Mary frequency shift keying system design in sea surface fluctuation. Xue Dandan, Kyu-Chil Park, Jihyun Park, and Jong R. Yoon (Pukyong National Univ., 45 Yongsoro Namgu, Busan 608-737, South Korea, jryoon@pknu.ac.kr)

The underwater acoustic channel can be expressed as a frequency selective fading channel due to multipath, sea surface roughness, and propagation medium change with time and space. Therefore the fading statistics varies with time, space, and frequency. Inherent factor of this fading change is related to a frequency dependent constructive or destructive interference change with time, space, and frequency. The magnitude of interference depends on closely on sea surface roughness. In this study, channel fading statistics are analyzed with delay spread and Doppler spread using pseudo noise (PN) signals. Based on this channel fading statistics, a low data rate M-ary frequency shift keying (MFSK) system of 1kbit/s, is designed for a command/control of navigation or an equipment status monitoring system. The forward error correcting code of convolution code is also applied. Experimental work is conducted in shallow water and it is found that the design parameters of MFSK system can be optimized using the measured fading statistics deduced from PN signal.

1aUW16. Differential pattern time delay shift coding underwater acoustic communication using parametric array. Jingwei Yin, Xiao Zhang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Rm. 801, ShuiSheng Bldg., Harbin, HeiLongJiang 150001, China, zhangxiao@hrbeu.edu.cn), and Yiming Zhou (ShangHai Acoust. Lab., Chinese Acad. Sci., ShangHai, China)

The long range and high data rate communication are two main targets of point to point underwater acoustic communication (PPUAC). In order to achieve the longer communication range, low frequency is the best choice. However, the data rate is limited by narrow available bandwidth at low frequency. So, the system band select must compromise between the data rate and communication range according to the application scenarios. Parametric array can achieve considerable bandwidth at the low frequency band with narrow beam pattern by nonlinear acoustics conversion. These characteristics meet the requirements of both data rate and communication range to a certain extent, and make the channel equalization more simple and effective. A PPUAC scheme based on differential pattern time delay shift coding using parametric array has been proposed. The system employs a 150 kHz primary frequency then get a 10 kHz secondary frequency wave. A proof test has been carried out under ice in SongHua River, HeiLongjiang Province, China. The error free communication was achieved in the trial. The communication scheme is expected to been used in underwater platform directional remote control and other scenarios.

Session 1pAAa

Architectural Acoustics and Noise: Uncertainty in Laboratory Building Acoustics Measurements

Matthew V. Golden, Chair
Pliteq, 616 4th Street, NE, Washington, DC 20002

Chair's Introduction—1:00

Invited Papers

1:05

1pAAa1. Evolution of ASTM E90 for measurement of sound transmission loss and related standards. Noral D. Stewart (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 101, Raleigh, NC 27607, noral14@sacnc.com)

The history ASTM E90 for measurement of sound transmission loss will be traced from the initial release of the first tentative standard in 1950, identifying significant changes that have occurred with each revision. To a lesser degree, the history of ASTM E413 on the sound transmission class, ASTM E1322 on rating outdoor-indoor sound transmission, ASTM E336 for measurement of sound isolation and insulation in the field, and ASTM E966 on outdoor to indoor measurements will be discussed.

1:25

1pAAa2. Understanding the concepts of uncertainty, reproducibility, and repeatability and the application to acoustic testing. Ralph T. Muehleisen (Energy Systems, Argonne National Lab., 9700 S. Cass Ave., Bldg. 202, Lemont, IL 60439, rmuehleisen@anl.gov)

The consideration of uncertainty in measurement has long been an important part of scientific measurement. As measurements became standardized and testing and measurement standards were developed, various forms of uncertainty were sometimes, but not always, considered. Understanding various forms of uncertainty and the basic concepts of uncertainty, reproducibility, and repeatability are important for testing even if the estimation of them is not directly part of the test standard. In this presentation, the concepts of uncertainty, reproducibility, and repeatability for acoustic testing are explained using the ASTM E90 test as an example for explanation.

1:45

1pAAa3. A review of academic research and data related to laboratory testing for sound transmission loss. Benjamin Shafer (Tech. Services, PABCO Gypsum, 3905 N 10th St., Tacoma, WA 98406, ben.shafer@quietrock.com)

The building noise control design and manufacturing industry currently relies heavily on laboratory-tested sound transmission loss data for the design and implementation of noise control treatments in buildings. Although most sound transmission loss laboratories achieve acceptable measurement repeatability, the reproducibility of this test data has recently fallen under scrutiny and is currently unknown for common building noise control partition designs. Key academic articles will be discussed that provide a fundamental understanding of how laboratory sound transmission loss measurement procedures and laboratory facilities may affect reproducibility. A review of comparative sound transmission loss data for various laboratories will also be provided in an effort to clearly illustrate why sound transmission loss testing reproducibility has come under scrutiny. It is essential that organizations for standards and design professionals become more aware of these realities.

2:05

1pAAa4. The influence of test fixture damping on the measurement of sound transmission loss. Jennifer A. Shaw, Charles T. Moritz, and Armando Carrera (Blachford, Inc., Blachford Acoust. Lab., 1445 Powis Rd., West Chicago, IL 60185, jshaw@blachfordinc.com)

The sound transmission loss (STL) provided by a material depends on factors such as its mass, damping, and stiffness. In the lab, the damping and stiffness of a panel is influenced by the test fixture. To reduce test to test and lab to lab variation, the test standards give some detail with regards to installing the test specimen and sealing around the edges to prevent leakage between test rooms. However, many times it is not possible to simulate typical installation and sealing methods due to the wide range of installation options for the product, and a heavy layer of clay or other material is used to seal the sample in the fixture. As the size of the sample decreases, the edge damping from clamping and/or sealing the sample can add significant damping to a panel. Too much damping can cause a significant overstatement of the sample STL. This paper examines the influence of edge damping due to sealing clay on the STL of steel and aluminum panels.

2:25

1pAAa5. Categories of repeatability and reproducibility in acoustical laboratories. John J. LoVerde and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

There are many categories of variability in acoustical laboratory testing. The repeatability of acoustical testing is generally defined based on repeated testing of the same assembly in the same laboratory and conditions over a short time period, while the reproducibility is based on testing a (nominally identical) assembly in a different laboratory. However there is a spectrum of uncertainties between these points. The authors previously documented "long term repeatability," the repeated testing of the identical specimen in the same laboratory with the same equipment and operator, but over an extended time period [J. Acoust. Soc. Am. **130**, 2355 (2011)]. The current study examines the "long term rebuild repeatability," a nominally identical assembly rebuilt in the same laboratory over a long time period. The results are compared with previously measured variability, and the implications for manufacturers and designers are discussed.

2:45–3:00 Break

3:00

1pAAa6. The National Voluntary Laboratory Accreditation Program and Acoustical Testing Services Laboratory Accreditation Program. Kari K. Harper (NIST NVLAP, NIST, 100 Bureau Dr., M/S 2140, Gaithersburg, MD 20899, kharper@nist.gov)

This presentation provides an overview of NVLAP, the National Voluntary Laboratory Accreditation Program of the National Institute of Standards and Technology (NIST), with particular focus on NVLAP's Acoustical Testing Services Laboratory Accreditation Program (LAP). Through the Acoustical Testing Services LAP, NVLAP provides third-party attestation of the conformance of its accredited acoustics laboratories to the requirements of ISO/IEC 17025:2005, *General requirements for the competence of testing and calibration laboratories*; NIST Handbook 150, *NVLAP Procedures and General Requirements*; and NIST Handbook 150-8, *NVLAP Acoustical Testing Services*. Specific topics covered in this presentation include a brief history of the Acoustical Testing Services LAP, the accreditation process, proficiency testing, and benefits of accreditation, such as international recognition of the competence of accredited laboratories and the concept of "tested once; accepted everywhere."

3:20

1pAAa7. What acousticians need to know about the third party sound transmission class testing process and acoustical materials performance validation. Eric P. Wolfram (Riverbank Acoust. Labs., 1512 S Batavia Ave., Geneva, IL 60134, ewolfram@alionscience.com)

The architectural product industry demonstrates sound transmission loss performance by commissioning third party tests from accredited laboratories performed according to the ASTM E90 standard. This presentation will provide insight into the materials testing process, laboratory quality management, and explanation of the report documents. The presenter will share experience and advice for the use of test results to evaluate acoustical product performance. Acousticians, architects, and engineers must play a role in improving quality of published data by demanding more complete information from manufacturer's marketing departments regarding product acoustical performance.

3:40

1pAAa8. Status report on the planned intra-laboratory study to improve American Society for Testing and Materials E90's precision and bias statement. Matthew V. Golden (Pliteq, 616 4th St., NE, Washington, DC 20002, mgolden@pliteq.com)

The ASTM E33 Committee on Building and Environmental Acoustics has recently undertaken an effort to improve the precision and bias statements of all of the standards under its purview. Based on an analysis of the current state of the precision and bias statements and each standards importance to the industry, ASTM E90 was chosen as the standard to focus on first. Improving the precision and bias statements requires an intra-laboratory performance study, also known as a round robin, to be performed. Previous round robins for ASTM E 90 used relatively low performing specimens, Sound Transmission Class (STC) 24-32. The current proposed round robin is planned to use specimens with a STC in the high 40's to low 50's. Details of these efforts will be shared.

Contributed Paper

4:00

1pAAa9. Reproducibility and repeatability of measuring noise level reduction using an artificial noise source. Rene Robert, Kenneth Cunefare (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Office 002, Atlanta, GA 30332, rrobert6@gatech.edu), Erica Ryherd (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE), and Javier Irizarry (School of Bldg. Construction, Georgia Inst. of Technol., Atlanta, GA)

Buildings that are subjected to aviation noise require extra sound isolation measures in order to keep the indoor noise levels below a certain threshold. The difference in sound pressure from outside to inside a building is often quantified as a single number known as the noise level reduction (NLR). Generally, the procedures described in ASTM E966-10 are followed in determining this number for a façade, which is then used to determine the modifications required. The standard allows testing with one of two noise sources: an actual traffic source or a loudspeaker. An investigation is underway to statistically evaluate the repeatability and reproducibility of measurements taken on a façade for a "test house" with a loud speaker. The test house is a single-room structure that was designed and constructed with materials and methods of a residence in a mixed-humid climate. The repeatability is defined as the ability for a specific test to be implemented multiple times with comparable results, while the reproducibility is the ability for various test configurations allowed within the standard to yield comparable results. The results of this analysis can be used to improve testing parameters, further validate the procedure, and provide an estimate of confidence bounds.

4:15–4:45 Panel Discussion

Session 1pAAb

Architectural Acoustics, Noise, and Structural Acoustics and Vibration: Session in Honor of Laymon Miller

Neil T. Shade, Cochair
Acoustical Design Collaborative, Ltd, 7509 Lhirondelle Club Road, Ruxton, MD 21204

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

Chair's Introduction—2:00

Invited Papers

2:05

1pAAb1. Family and life experiences with Laymon. Robert L. Miller (Harris Miller Miller & Hanson Inc., 77 South Bedford St., Burlington, MA 01803, rmiller@hmmh.com)

This talk will bring to light memorable personal recollections growing up with Laymon as husband, father, and mentor. My own memories will illustrate how his love of acoustics, the people he worked with, and the problems he worked on ultimately led to my own career in acoustics. They will include a recounting of childhood business trips, flying experiences, and a summer job opportunity at Bolt Beranek and Newman. The talk also will include his wife, Lucy's, still-vivid recollections of Laymon's first years at the Harvard Underwater Sound Lab, his nine years at Penn State, and many years later their times together as she accompanied him on his BBN lecture series. Unpublished excerpts from Laymon's five-volume autobiography will be used to illustrate his own detailed and insightful accounting of other family and life experiences, including his own "ethical will," written for later generations to better understand and appreciate the values and beliefs for which he most wanted to be remembered.

2:25

1pAAb2. Laymon Miller, distinguished consultant in noise control engineering at Bolt Beranek and Newman. Leo Beranek (Consultant in Acoust., 10 Longwood Dr., Westwood, MA 02090, beranekleo@ieee.org)

Laymon Miller was employed in Harvard's Underwater Sound Lab (HUSL), under the direction of Professor F. V. Hunt during World War ll. When that laboratory closed, Laymon joined the consulting staff at Bolt Beranek and Newman (BBN). Laymon became BBN's leading expert on the reduction of noise and vibration in ventilating systems. One of his important jobs was noise reduction at New York's Lincoln Center's Philharmonic Hall. He also became expert in vibrations transmitted through the earth. In particular, he designed the vibration isolation pads used to reduce the transmission of railroad track vibrations into adjacent buildings in a number of cities, including Boston's Back Bay and Montreal. Practically every important building acoustics project that came into BBN received his attention. He gained the reputation of more chargeable consulting time each month than any other employee. Besides mentoring numerous new BBN employees he taught highly sought after courses on noise reduction for architects, building, industrial and plant engineers, as well as for other acoustical consultants.

2:45

1pAAb3. Reminiscences on Laymon Miller's remarkable twenty seven year consulting career at Bolt Beranek and Newman in Cambridge. William J. Cavanaugh (Cavanaugh Tocci Assoc. Inc., 3 Merifield Ln., Natick, MA 01760-5520, wcavanaugh@cavtocci.com)

Laymon Miller joined the noise control engineering consulting staff of Bolt Beranek and Newman (BBN) on August 1, 1954, bringing an impressive record of experience in acoustics from Harvard University's top secret Underwater Sound Lab (HUSL) during WWll followed by ten years as head of the acoustics section at Penn State's Ordinance Research Lab (ORL). Laymon provided invaluable mentoring for many new BBN consulting staff members like this author whose only experience in acoustics were introductory courses in architectural acoustics taught by Physics Professor Richard Bolt at MIT's School of Architecture and the MIT Acoustics Lab, Laymon quickly established a reputation among BBN employees and clients alike as an outstanding teacher and contributed to BBN's unofficial title as the "third important graduate school in Cambridge." He documented his diverse consulting career through numerous technical papers and articles, through technical brochures he prepared for clients like the US Army Corps of Engineers,manufactures like the Baltimore Air Coil Company and through reports he prepared for thousands of BBN projects on which he served as principal consultant. Laymon retired from BBN in 1981 after a remarkable 27 year career making quieter buildings, concert halls, workplaces, communities, transportation vehicles, and in general, "the world a better place."

3:05

1pAAb4. Laymon N. Miller—Contributions to industrial and community noise control. Eric W. Wood (Acentech, 33 Moulton St., Cambridge, MA 02138, ewood@acentech.com)

Laymon N. Miller, following graduation in 1939 from the University of Texas in Austin, spent time first at the Underwater Sound Lab at Harvard University and then at Penn State as Head of the Acoustics Section of Ordnance Research Lab (ORL). In 1954, he joined Bolt Beranek and Newman where he was honored as their first Principal Consultant. This presentation describes his contributions to industrial and community noise control while consulting for a wide range of many clients for 27 years, until his retirement in 1981. Laymon Miller, a gentleman, friend, and colleague.

3:25

1pAAb5. Origin and history of the Laymon Miller noise course. Reginald H. Keith (Hoover & Keith Inc., 11381 Meadowglen Ln., Ste.. I, Houston, TX 77082, reggie.keith@hoover-keith.com)

For 20 years, starting in 1969, Laymon Miller produced and taught a seminal course in noise and vibration control in many different venues and settings. In this paper, I will present information as to the origins of this course, my recollections of first attending the course in 1977, and our experiences in the continuation of the course since 1989.

3:45

1pAAb6. Laymon Miller—An exemplary acoustical consultant. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooksacoustics.com)

Laymon Miller was a “Consultant’s consultant.” He embodied a wonderful example of how to conduct oneself in the engineering consulting business, providing leadership in defining the functions and responsibilities of acoustical consultants. As a leader of our profession, Laymon was also a great friend to the National Council of Acoustical Consultants (NCAC). Former NCAC Newsletter Editor Bill Cavanaugh asked Laymon if he would serve as a guest editor for a continuing series tentatively titled “War stories … from the Consulting Veteran’s files.” Laymon answered the call, and this series continued for many years, capturing the “priceless gems” from which all of us, at all experience levels, can learn about the problems faced by consultants, and importantly about the solutions as applied in the field. Laymon was a talented and generous teacher, in print and in person, of those inside and outside of acoustical consulting. His experiences, related with insight and humor, have provided guidance to generations of those in general industry and to acoustical consulting practitioners alike. Laymon was elected an Honorary Member of NCAC in 1993, received the NCAC C. Paul Boner Award in 2007 and was presented the Institute of Noise Control Engineering (INCE) Outstanding Educator Award in 2008.

4:05

1pAAb7. Experiences as editor of Laymon Miller’s book *An NCAC Anthology in Noise and Vibration*. Neil T. Shade (Acoust. Design Collaborative, Ltd., 7509 LHirondelle Club Rd., Ruxton, MD 21204, nts@akustx.com)

This presentation will recount the experiences of serving as editor of Laymon Miller’s book, *An NCAC Anthology in Noise and Vibration*, published in 2013 by the National Council of Acoustical Consultant’s (NCAC). Ever active in retirement, and at the behest of William Cavanaugh, Laymon contributed 60 papers from 1996 to 2012 to the NCAC *Newsletter*. These papers, along with his industry publications from 1957 to 2008, were the basis for his book. A committee was formed within NCAC to oversee the compilation and production of his book with this author serving as editor. The “engineer” in Laymon was evident in the meticulous organization of his writings and instructions provided to the editor. *An NCAC Anthology in Noise and Vibration* is organized in two parts. The first contains industry publications, arranged by topic. The second is Laymon’s NCAC articles arranged chronologically. The book concludes with his autobiography, which he humbly described “as a life of surprises.” Laymon dedicated his book to his friend and colleague Leo Beranek. Through editing the book the author gained insight into Laymon’s seminal contributions to building acoustics, noise and vibration control, community noise, his acoustic consulting adventures, and the work ethic of this remarkable man.

Session 1pABa

Animal Bioacoustics: Physiology of Animal Hearing

Edward J. Walsh, Chair
Research, Boys Town Natl. Res. Hospital, 555 North 30th Street, Omaha, NE 68131

Contributed Papers

1:00

1pABa1. Discrimination of frequency-modulated sweeps by laboratory mice. Laurel A. Screven and Micheal L. Dent (Univ. at Buffalo, B29 Park Hall, Buffalo, NY 14260, laurelsc@buffalo.edu)

Mice often produce ultrasonic vocalizations (USVs) that sweep upwards in frequency from around 60 kHz to around 80 kHz, and similarly sweep downwards in frequency from 80 kHz to 60 kHz. Whether or not these USVs are used for communication purposes is still unknown. Determining the ability of mice to discriminate between synthetic upsweep and downsweep frequency-modulated stimuli will expand the current knowledge about acoustic communication in mice. Mice were trained and tested using operant conditioning procedures and positive reinforcement to discriminate between upsweeps and downsweeps. The stimuli varied in bandwidth, duration, and direction of the sweep. The animals responded when they heard a change in the repeating background, indicating that they could discriminate background from target. The mice performed significantly worse discriminating between background and targets when the stimuli occupied the same bandwidths. Further, the mice's discrimination performance became much worse when the duration approached that of their natural vocalizations. When the sweeps occupied different frequency ranges and longer durations, discrimination performance improved. These results collected using artificial stimuli created to mimic natural USVs indicate that the bandwidth of vocalizations may be much more important for communication than the frequency contours of the vocalizations. [Work supported by NIH DC012302.]

1:15

1pABa2. Hearing sensitivity in the Greater Prairie Chicken (*Tympanuchus cupido*). Edward J. Walsh (Developmental Auditory Physiol. Lab, Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, edward.walsh@boystown.org), Cara Whalen, Larkin Powell, Mary B. Brown (School of Natural Resources, Univ. of Nebraska, Lincoln, NE), and JoAnn McGee (Developmental Auditory Physiol. Lab, Boys Town National Res. Hospital, Omaha, NE)

As their scientific name implies, Greater Prairie Chickens (*Tympanuchus cupido*) are well known for their vocalizations, particularly those produced by males during courtship. In this report, the auditory sensitivity of a small cohort of Greater Prairie Chickens inhabiting the grasslands of southeastern Nebraska is presented as part of an effort to assess the capacity of wind turbine farm generated noise to mask male courtship vocalizations. Birds were captured from breeding grounds (leks) between March and June of 2014 and sensitivity to tone-burst stimuli was assessed using the auditory brainstem response. While response waveforms were typical of responses observed in other avian species, as was the bandwidth of audibility curves, preliminary data suggest the possibility that Greater Prairie Chickens may be unusually sensitive to low frequency tone-bursts in the vicinity of the dominant spectral component of the "booming" call, a prominent, intense vocalization that males produce during courtship. As part of the larger study that is designed to evaluate the potential impact of wind turbine farm operation on reproductive success, demography and overall survival rates, the potential masking influence of noise produced by turbine operations at wind farms will be discussed in relation to sensitivity findings.

1:30

1pABa3. Place specificity of dolphin auditory evoked potentials assessed with high-pass masking noise. James J. Finneran (SSC Pacific Code 71510, US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Jason Mulsow, Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA), and Robert F. Burkard (Univ. of Buffalo, Amherst, New York)

Auditory evoked potential measurements are commonly used for hearing assessment in marine mammals for which psychophysical testing is not practical. Evoked potential measurements typically employ electrical signals consisting of short-duration tone-pips or sinusoidal amplitude modulated tones, which are then presented to piezoelectric underwater sound projectors. Although tone-pip and short-duration tonal stimuli may possess relatively narrow frequency bandwidth, the resulting physiological responses may possess much greater bandwidth, especially for lower frequency stimuli at higher stimulus levels. In this study, high-pass masking noise techniques were used to examine the place specificity of auditory evoked responses from click, tone-pip, and sinusoidal amplitude modulated tones in bottlenose dolphins (*Tursiops truncatus*). The experimental methods for generating and spectrally equalizing masking noise and click stimuli will be presented, along with the effect of compensated clicks with uncompensated clicks and ABR latencies and amplitudes. [Funded by U.S. Navy Living Marine Resources Program.]

1:45

1pABa4. Equal-latency curves and auditory weighting functions for bottlenose dolphins (*Tursiops truncatus*) and California sea lions (*Zalophus californianus*). Jason Mulsow (National Marine Mammal Foundation, 2240 Shelter Island Dr., Ste. 200, San Diego, CA 92106, jason.mulsow@nmmf.org), James J. Finneran (U.S. Navy Marine Mammal Program, SSC Pacific, Code 71510, San Diego, CA), and Carolyn E. Schlundt (ITT Exelis Corp., San Diego, CA)

Reaction time (RT) data obtained from simple tonal detection tasks have been used to estimate frequency-specific equal-loudness contours in non-human animals. In order to guide the design of auditory weighting functions for marine mammals, equal-latency contours were generated using RT data from a simple tonal detection including two bottlenose dolphins (under water) and three California sea lions (in air). Median RT increased exponentially with decreased SPL in all cases. Equal-latency contours for near-threshold RTs were similar to audiograms in both species. Data for the sea lions showed some compression of equal-latency contours with increases in SPL; however, large inter-subject differences in the data for dolphins made results for that species more difficult to interpret. The equal-latency contours for all subjects progressively diverged from predicted equal-loudness contours at higher SPLs, likely a result of very small changes in RT with relatively large increases in SPL. As a result, the contours of most interest for designing weighting functions for high-level noise exposures were also the least reliable. The general similarity of most of the contours to species-typical audiograms suggests that more easily obtained auditory thresholds may provide useful approximations for weighting. [Funded by U.S. Navy Living Marine Resources Program.]

Session 1pABb

Animal Bioacoustics: Advances in Processing Bioacoustic Data

Kristin B. Hodge, Chair
Bioacoustics Research Program, Cornell University, 159 Sapsucker Woods Road, Ithaca, NY 14850

Contributed Papers

2:45

1pABb1. Continental scale acoustic monitoring program: One year of data. Samuel L. Denes (Biology, Syracuse Univ., 116 Appl. Sci. Bldg., University Park, PA 16802, sld980@psu.edu), Susan E. Parks, Leanna Matthews, Hannah Blair (Biology, Syracuse Univ., Syracuse, NY), Pramod Varshney (Elec. Eng. & Comput. Sci., Syracuse Univ., Syracuse, NY), and Kurt Fristrup (Natural Sounds and Night Skies Div., National Park Service, Fort Collins, CO)

A multiyear project is underway to demonstrate the benefits of incorporating acoustic monitoring into the National Ecological Observatory Network (NEON). The NEON project seeks to generate data for the study of continental scale phenomena. Acoustic recordings can be used to determine the presence of acoustically active biota without the presence of field technicians. We have deployed stand-alone acoustic recorders at four NEON sites. Data from these recorders can be used to document spatio-temporal shifts in the presence of acoustically active species of birds, anurans, and insects. Amplitude, frequency band energy, and statistical detection methods have been compared to demonstrate the implementation of automated detection algorithms. Results from acoustic biota surveys are compared with traditional biota surveys. Comparisons of the relative contributions of acoustic energy from geophysical, biotic, and anthropogenic sources within and between sites will be examined. Preliminary results from the first year and a half of acoustic data collection will be presented. [Project supported by NSF award #1340669.]

3:00

1pABb2. Improving knowledge and understanding of underwater sound from exploration and production activities and potential interaction with marine life: An overview of a research program. Ian Voparil (IOGP E&P Sound & Marine Life Joint Industry Programme (JIP), P.O. Box 61933, Rm. 3412, New Orleans, LA 70161, ian.voparil@shell.com)

The International Oil and Gas Producers E&P Sound & Marine Life Joint Industry Programme (JIP), a partnership of 13 oil and gas producing companies, is the world's largest non-governmental funder of research to increase understanding of the potential effects of E&P sound on marine life. The JIP provides objective, scientific information that: Informs and updates policy decision makers and regulatory development processes that affect E&P operations globally Determines the basis for mitigation measures that are protective of marine life, cost effective, and credible with outside stakeholders feeds into planning for efficient E&P project development that is environmentally protective. Since 2006, it has spent $25 million on research and in 2013, the program embarked on a new phase with an additional budget of $18 million. The JIP regularly consults with regulators regarding their needs for data, to enable development of fact-based regulatory decisions. The program co-funds selected projects with government agencies. Research is conducted by independent scientists who are encouraged to publish their results in peer-reviewed literature. To date, JIP-funded research has been published in 75 peer-reviewed papers and all final project reports publically available. During this talk, results of the previous JIP-funded research will be presented. Furthermore, an overview of ongoing and future studies is provided.

3:15

1pABb3. Acoustic monitoring of Bryde's whales (*Balaenoptera edeni*) in the northern Gulf of Mexico. Kristin B. Hodge, Jamey T. Tielens, and Aaron N. Rice (BioAcoust. Res. Program, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, kbhodge@cornell.edu)

Bryde's whales (*Balaenoptera edeni*) are the most common mysticete species occurring in the Gulf of Mexico, yet due to a dearth of visual sightings and stranding data, little is understood regarding their spatial distribution and seasonal occurrence patterns. Two putative call types unique to this population of Bryde's whales have recently been identified, providing an opportunity to understand Bryde's whale spatial and temporal occurrence patterns using passive acoustic monitoring methods. In order to evaluate Bryde's whale presence in the area, marine autonomous recording units (MARUs) were deployed from June 2010 through September 2011 in the northern Gulf of Mexico where Bryde's whales have been historically observed. Distribution, seasonality, and diel patterns were examined separately for each of the call types at all sites and days during which acoustic data were collected by the MARUs. Preliminary results show Bryde's whales were acoustically detected throughout the recording period. Of the two call types associated with Bryde's whales, one call type was produced consistently more often than the other; however, both call types showed a similar diel pattern. These results suggest Bryde's whales may reside or visit this area year-round, which may facilitate future monitoring efforts for this poorly studied population.

3:30

1pABb4. Automatic manatee count using passive acoustics. Jorge M. Castro, Arturo Camacho, and Mario Rivera (CITIC, Universidad de Costa Rica, Ciudad Universitaria Rodrigo Facio, San Pedro, San José 11501, Costa Rica, pctreepkfloyd@gmail.com)

The West Indian manatee is a threatened species throughout its known range. To improve its conservation, it is necessary to locate and count the individuals. This is a very difficult task due to the cryptic behavioral characteristics of the species and the environmental constrains, particularly in the Central American region, where muddy waters produce limited visibility. A method to estimate manatee population through vocalizations is a reliable, inexpensive, non-invasive, and novel option in the region. Digital signal processing techniques are proposed to identify individuals in field recordings of manatee vocalizations. The proposed methods for signal denoising and vocalization recognition have produced promising results. Further work needs to be done to identify individuals.

3:45–4:00 Break

4:00

1pABb5. Localization of individual call among croakers' chorus using a stereo recording system. Masanori Ito, Ikuo Matsuo (Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, matsuo@cs.tohoku-gakuin.ac.jp), Tomohito Imaizumi, and Tonomari Akamatsu (National Res. Inst. of Fisheries Eng., Fisheries Res. Agency, Kamisu, Japan)

Passive acoustic monitoring (PAM) is widely used in cetacean census in the ocean. In these years, PAM has been applied for various species out of mammals such as fish and crustaceans. For detection, classification and density estimation in PAM, isolation of each biological sound and reliable detection is essentially needed. Choruses of fish sounds were recorded by a stereo sound monitoring system on the seafloor in off Choshi, Chiba prefecture, Japan in May 2013. Numbers of croaker sounds were recorded and the simple separation in time domain was not possible in chorus situation. A blind source separation method based on independent component analysis was applied to separate mixed sounds in order to obtain individual sounds and to localize them. The directions of sound arrivals could be estimated by using the cross correlation function of the separated signals and the estimated directions were plotted. Using proposed methods, individual call sounds could be localized and inter-pulse intervals could be accurately estimated. Using separated sequence of sounds from an individual, the movement of phonating fish could be measured using PAM. [This study was supported by CREST, JST.]

4:15

1pABb6. Newtonian and weighted essentially non-oscillatory models to describe the generation and propagation of the Cicada mating calls. Derke Hughes, Sheri L. Martinelli (NUWCDIVNPT, 1176 Howell St., Newport, RI 02841, derke.hughes@verizon.net), Allan D. Pierce (Mech. Eng., Boston Univ., East Sandwich, MA), Richard A. Katz, and Robert M. Koch (NUWCDIVNPT, Newport, RI)

Experiments and analyses of Hughes *et al.*, JASA, 2009 are the origins of this research where we study the in-air waveform generation and propagation of the acoustic signals generated by cicadas. The sound generation is studied in a Newtonian model and the sound propagation is analysis by a numerical solver for viscous Burgers' equation. The time histories from the tymbal surface velocities recorded by a laser Doppler vibrometer to the microphones positioned near the cicadas provide the test data. The Newtonian model describes the sound production systems process to generate the mating call signal structure. The numerical solver employs weighted essentially non-oscillatory (WENO) reconstruction to approximate the first and second derivatives of the semi-discrete operator. The WENO is utilizes due to the non-smooth structure of the cicada propagating waveform. Principally, the cicada mating signal in question has sharp transitions, since spectral methods tend to produce spurious oscillations as a result of attempting to represent a discontinuous function by a Fourier basis expansion. Thus, these analytical models are computationally tested to determine if the results capture the sound production and the transmission of the cicada mating calls. To verify the models are meaningful, the simulations are verified with real experimental data.

4:30

1pABb7. Simple mechanical model reproduces complex calls in a fish vocal system: Implications for the evolution of vertebrate acoustic communication systems. Aaron N. Rice (BioAcoust. Res. Program, Cornell Lab of Ornithology, Ithaca, NY), Sang Min Han, Bruce R. Land (School of Elec. and Comput. Eng., Cornell Univ., 8452 Low Rise 8, Ithaca, NY 14853, sh833@cornell.edu), and Andrew H. Bass (Dept. of Neurobiology and Behavior, Cornell Univ., Ithaca, NY)

A mathematical model has been developed for vocalizations of the three-spined toadfish (*Batrachomoeus trispinosus*) and the plainfin midshipman (*Porichthys notatus*); acoustically active fish species that have served as model systems for studying acoustic communication mechanisms among vertebrates. The toadfish has a vocal organ comprised of a pair of bilaterally separated swim bladders that generate acoustic nonlinearities similar to those found in tetrapods. Midshipman have one swim bladder with sounds lacking the nonlinearities. We modeled the toadfish mechanism as a system of two harmonic oscillators coupled with a tendon-like string, and the midshipman one by the same system without the coupling. The coupling in the toadfish model, along with nonlinear oscillator position and velocity dependent terms, generates signatures of nonlinearity such as deterministic chaos and bifurcating harmonics. The system generates sounds that simulate naturalistic calls of both toadfish and midshipman depending on the input parameters. We built an optimizer to minimize the difference between the power density spectra of the simulated call and an empirical recording. We successfully computed optimal toadfish and midshipman calls and simulated transection experiments, together leading to a deeper understanding of the diversity of vertebrate vocalization mechanisms. [Research support from Rawlings Cornell Presidential Scholars and NSF IOS 1120925.]

Session 1pPA

Physical Acoustics: Acoustofluidics: Interaction of Acoustics and Fluid Dynamic Phenomena

Charles Thompson, Cochair
ECE, UMASS, 1 University Ave., Lowell, MA 01854

Max Denis, Cochair
University of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854

Chair's Introduction—1:00

Invited Papers

1:05

1pPA1. Acoustic tweezers: Manipulating particles, cells, and fluids using sound waves. Tony Jun Huang (Eng. Sci. and Mech., Penn state Univ., Millennium Sci. Complex, PSU, N-330, University Park, PA 16802, junhuang@engr.psu.edu)

The ability to manipulate cells, micro/nanoparticles, and fluids in a biocompatible and dexterous manner is critical for numerous biological studies and applications such as cell-cell communication, biosensing, tissue engineering, regenerative medicine, and lab on a chip. Here, we summarize our recent progress on an "acoustic tweezers" technique that utilizes acoustic waves to manipulate particles, cells, organisms, and fluids. This technique is capable of manipulating cells and microparticles regardless of shape, size, charge, or polarity. Its power intensity, approximately 10^7 times lower than that of optical tweezers, compares favorably with those of other active patterning methods. Cell viability, proliferation, and gene expression have revealed the technique to be extremely biocompatible. The aforementioned advantages, along with this technique's simple design and low-cost, compact design, render the "acoustic tweezers" technique a promising tool for various applications in biology, chemistry, engineering, biophysics, and materials science.

1:30

1pPA2. Intense cavitation in microfluidics for bio-technology applications. Siew-Wan Ohl, Tandiono Tandiono, Evert Klaseboer (Inst. of High Performance Computing, 1 Fusionopolis Way, #16-16 Connexis North, Singapore 138632, Singapore, ohlsw@ihpc.a-star.edu.sg), Dave Ow, Andre Choo (Bioprocessing Technol. Inst., Singapore, Singapore), and Claus-Dieter Ohl (School of Physical and Mathematical Sci., Nanyang Technolog. Univ., Singapore, Singapore)

This study reports the use of intense ultrasonic cavitation in the confinement of a microfluidics channel [1], and the applications that has been developed for the past 4 years [2]–[5]. The cavitation bubbles are created at the gas-water interface due to strong capillary waves which are generated when the system is driven at its natural frequency (around 100 kHz) [1]. These bubbles oscillate and collapse within the channel. The bubbles are useful for sonochemistry and the generation of sonoluminescence [2]. When we add bacteria (*Escherichia coli*), and yeasts (*Pichia pastoris*) into the microfluidics channels, the oscillating and collapsing bubbles stretch and lyse these cells [3]. In another application, human red blood cells are added to a microchamber. Cell stretching and rapture are observed when a laser generated cavitation bubble expands and collapses next to the cell [4]. A numerical model of a liquid pocket surrounded by a membrane with surface tension which was placed next to an oscillating bubble was developed using the Boundary Element Method. Lastly, new results on gene transfection [5], lysing of bacterial spores, and emulsification by ultrasonic bubbles will be presented. References: [1] Tandiono *et al*., Lab Chip **10**(14), 1848–1855 (2010). [2] Tandiono *et al*., Proc. Natl. Acad. Sci. U.S.A. **108**(15), 5996–5998 (2011). [3] Tandiono *et al*., Lab Chip **12**(4), 780–786 (2012). [4] Tandiono *et al*., Soft Matter **9**(36), 8687–8696 (2013). [5] Ling *et al*., Biotechnol. J. **9**(8), 1081–1087 (2014).

1:55

1pPA3. Steady streaming around a pulsating bubble located at the velocity node of a standing wave. Mohammad AlHamli and Satwindar S. Sadhal (Aerosp. & Mech. Eng., Univ. of Southern California, Olin Hall OHE 430, Los Angeles, CA 90089-1453, alhamli@usc.edu)

We have examined the effect of the no-slip boundary condition on the steady streaming around a radially pulsating bubble, located at the velocity node of a standing sound wave. The no-slip condition can take place on the bubble surface under many circumstances, especially if the surface is contaminated or otherwise coated. We applied the singular perturbation analysis with the oscillation amplitude, $\varepsilon = U_0/(a\omega) << 1$, as a small parameter and took the frequency parameter, $M^2 = i\omega a^2/\nu$, to be large. Here, a, U_0, ω and ν are length scale, velocity scale, frequency, and kinematic viscosity, respectively. We further assumed that the bubble stays spherical in shape, the wavelength is significantly larger than the bubble radius, and no fluid transport takes place from the bubble to the surroundings. Additionally, the lateral and the radial oscillations, while at the same frequency, have a phase difference φ, and the amplitudes of both the oscillations are small compared to the bubble radius. We found that the streaming happened at a lower order than the non-pulsating case and it is more intense than the case of the shear free boundary condition. Furthermore, the phase difference was found to play a significant role in the streaming flow direction and its intensity.

2:20

1pPA4. Surface acoustic wave microfluidics: Thin films and drop splitting. James Friend (Mech. and Aerosp. Eng., Univ. of California, San Diego, 345F Structural and Mech. Eng., M.S. 411 Gilman Dr, La Jolla, CA 92093, jfriend@eng.ucsd.edu)

A renaissance in surface acoustic waves (SAW) has occurred, due mainly to their enormously powerful ability to manipulate fluids and colloids for microfluidics. Beyond the routine manipulation of drops, mixing, and separation and concentration of colloids using SAW, we have found fascinating behavior in the formation and propulsion of thin fluid films from a few tens to less than a micrometer thick in various directions, either with or against the acoustic wave propagation. Furthermore, we have seen—and been able to control through careful input conditions—the splitting of fluid drops. Both have a broad range of applications that will be briefly discussed, but more important is the underlying physics that illustrates the rich interaction of the acoustics, fluid dynamics, and free fluid interface in these systems.

2:45

1pPA5. Transition to turbulence in acoustically driven flows. Vineet Mehta (MIT Lincoln Lab., 244 Wood St., Lincoln, MA 02420-9108, vineet@ll.mit.edu), Charles Thompson, Kavitha Chandra (Univ. of Massachusetts Lowell, Lowell, MA), and Max Denis (Mayo Clinic, Rochester, MN)

Direct numerical solutions of the three-dimensional time-dependent Navier-Stokes equation are presented for the evolution of three-dimensional finite-amplitude disturbances in the Stokes boundary layer. The basic flow is driven harmonically in time and the disturbances represent a departure from the basic state of the fluid. For fixed value of the fluid viscosity, instability is shown to be a function of the amplitude of oscillation of the basic flow, excitation frequency, and the channel wall geometry. Special attention is given to pressure gradient and surface wave driven flows. Conditions for instability and transition are outlined.

3:10–3:25 Break

Contributed Papers

3:25

1pPA6. Examples of viscous phenomena relevant to second-order responses to ultrasound. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu) and Likun Zhang (Phys. Dept., Univ. of Texas, Austin, TX)

It has long been realized that viscosity strongly affects the response of drops and bubbles to the radiation pressure of modulated ultrasound. See for example an analysis of the viscous damping of bubble and drop shape oscillations and related experiments [T. J. Asaki and P. L. Marston, J. Fluid Mech. **300**, 149–167 (1995)]. There has been some recent interest in the importance of the viscosity of the fluid surrounding objects of interest when considering other second-order responses. These include the following quantities or responses associated with scattering by spheres and other objects: radiation force of progressive plane waves and Bessel beams [P. L. Marston, Proc. Meet. Acoust. **19**, 045005 (2013)]; power extinction of acoustic beams [L. Zhang and P. L. Marston, Bio. Opt. Express **4**, 1610–1617 (2013); (E) 4, 2988 (2013)]; and the radiation torque of vortex beams and orthogonal standing waves [L. Zhang and P. L. Marston, J. Acoust. Soc. Am. **131**, 2917–2921 (2014)]. The underlying assumptions and limitations of some current and prior approaches will be noted along with some applications to acoustophoresis. [Work supported by ONR (Marston) and by the 2013-14 F. V. Hunt Postdoctoral Fellowship (Zhang).]

3:40

1pPA7. Understanding the fluid dynamics associated with macro scale ultrasonic separators. Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA), Kedar Chitale, Walter Presz (FloDesign Sonics, 380 Main St., Wilbraham, MA 01095, k.chitale@fdsonics.com), and Olivier Desjardins (Mech. and Aerosp. Eng., Cornell Univ., Ithaca, NY)

Acoustic standing wave fields are widely used in MEMS applications to separate micron sized particles from fluids. However, the use and understanding of macro scale ultrasonic separators are still limited and challenging. These systems rely on acoustic radiation forces for trapping and clumping of dispersed phase particles. The clumps of particles then continuously separate out due to enhanced gravity or buoyancy in a flowing system. Typical flow Reynolds numbers are less than 50, particle concentrations up to 20%, ultrasonic standing wave fields at frequencies of 2 MHz, and acoustic pressure amplitudes of about 1 MPa. At such small Reynolds numbers, the flow is dominated by shear forces and the drag on clumps of particles is significantly lower than Stokes drag on a single particle. The fluid dynamics associated with these systems is extremely complex due to the coupling between the fluid flow field, suspended particles, and acoustic radiation forces. This work discusses the key physics involved and explains our current understanding of operation of macro scale acoustic separators. The status of CFD efforts to predict the flow fields and particle clumping in such systems is presented and compared to experimental results.

3:55

1pPA8. Experimental and numerical acoustofluidics in bulk acoustic wave devices at ETH Zurich. Philipp Hahn, Ivo Leibacher, Andreas Lamprecht, Peter Reichert, and Jurg Dual (Inst. of Mech. Systems (IMES), ETH Zurich, Tannenstrasse 3, Zurich CH-8092, Switzerland, hahnp@ethz.ch)

Ultrasonic fluid cavity resonances in acoustofluidic micro-devices can be exploited to miniaturize important operations for the handling of beads, cells, droplets, and other particles. With a growing number of experimentally tested unit operations, acoustofluidics holds increasing promise for emerging applications in bio- and microtechnology on lab-on-a-chip systems. We provide an overview of our research activities during the last years with a focus on the latest experimental setups and advances in the numerical simulation. Specifically, we present micro-devices with impedance matched cavity walls that allow a more flexible device design. Further, we show devices for the handling of fluid droplets and report on a method for the direct measurement of the acoustic radiation force on micro-particles. Due to the rapidly growing computational capabilities, numerical simulation has become a valuable tool in acoustofluidics research. We present a numerical model that accurately mimics the boundary layer damping inside the fluid cavity, allowing to make predictions of the attainable acoustic amplitudes and radiation forces. Furthermore, we demonstrate how numerical optimization of the device geometry is used to design new devices in an automatic fashion. Finally, we show how the radiation forces and torques can be deduced from the simulated acoustic fields to compute trajectories of complex shaped particles.

4:10

1pPA9. Inhibition of Rayleigh-Bénard convection via oscillatory acceleration. Anand Swaminathan, Steven L. Garrett (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, aswaminathan@wesleyan.edu), and Robert W. Smith (Appl. Res. Lab., The Penn State Univ., State College, PA)

The ability to dynamically stabilize Rayleigh-Bénard convection by imposition of sinusoidal acceleration is of interest to groups who design and study thermoacoustic machines, as the introduction of unwanted convection can have deleterious effects on the desired operation and efficiency of the machine. These performance issues, tentatively attributed to convective instability, have been observed both in traveling wave thermoacoustic refrigerators and in pulse tube cryocoolers. This presentation discusses an ongoing experiment designed to determine the vibratory conditions under which a small, rectangular container of statically unstable fluid may be stabilized by vertical vibration, to test the computational methods of R. M. Carbo [J. Acoust. Soc. Am. **135**(2), 654–668 (2014)]. Measurement methods developed to determine the onset and inhibition of convection will be discussed. These include the measurement of heat transport employing a feedback thermal control loop and direct optical observation using a lightsheet produced by a laser diode source illuminating seeded particles. Preliminary results in both the static and vibratory conditions will be presented. [Work supported by the ARL Walker Graduate Assistantship, the Office of Naval Research, and ARPA-E.]

4:25

1pPA10. Signal coherence of broadband sound propagation through a refractive and turbulent atmosphere. Jericho E. Cain, Sandra L. Collier (U.S. Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, jericho.cain@gmail.com), Vladimir E. Ostashev, and David K. Wilson (U.S. Army Engineer and Development Ctr., Hanover, NH)

Atmospheric turbulence and refraction have significant effects on acoustic signals. The signal coherence can be directly measured and can yield information for use in source localization and classification; however, the effects of turbulence and refraction decrease the accuracy of such efforts. The signal coherence of a broadband acoustic signal above an impedance ground in a refractive, turbulent atmosphere with spatial fluctuations in the temperature and wind velocity is modeled and the results for several atmospheric cases are considered.

4:40

1pPA11. The use of sound speed in downhole flow monitoring applications. Haldun Unalmis (Weatherford, 22001 North Park Dr., Kingwood, TX 77339, haldun.unalmis@weatherford.com)

This paper describes the use of sound speed in flow monitoring applications in the high-pressure/high-temperature downhole environment. Downhole flow monitoring is an area that continuously receives attention for many reasons including zonal production allocation in multi-zone intelligent completions with inflow control valves (ICV), detection of production anomalies, as well as reduction of surface well tests and facilities. The propagation speed of a sound wave is a powerful tool to extract useful information from a flowing fluid medium in pipe whether the medium consists of a single-phase or multiphase flow. Considering the complex nature of the flow patterns and changing phase fractions from reservoir to surface, obtaining the propagation speed of sound in this harsh environment is not a trivial task, especially if the interest is real-time flow monitoring. The demanding applications span a wide spectrum from very noisy medium (usually created in gas/liquid flows by the presence of ICV) to very quiet medium, which usually originates from slow-moving liquid/liquid flows. Real-life examples are used for demonstrations. Although most examples are based on strain-based local sensing of the flow, the use of sound speed is independent of the methodology and can be implemented by other methods such as acoustic-based distributed sensing.

4:55

1pPA12. Numerical investigation of aeroacoustic characteristics of a circular cylinder in cross-flow and in yaw. Xiaowan Liu, David J. Thompson (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton SO17 1BJ, United Kingdom, xl25g11@soton.ac.uk), Zhiwei Hu (Univ. of Southampton, Southampton, United Kingdom), and Vincent Jurdic (Arup Acoust., Winchester, United Kingdom)

It is well known that aerodynamic noise becomes dominant when the speed of high-speed trains exceeds about 300 km/h. The pantograph is one of the main aerodynamic noise sources, particularly in the presence of noise barriers which shield the sources on the lower part of the train more effectively. The pantograph consists of a number of slender bodies with different cross-sections. In the current research, the aerodynamic characteristics of a circular cylinder have been investigated through computational fluid dynamics simulations using a Delayed Detached-Eddy Simulation model. Then the aeroacoustic behaviour has been predicted by using Ffowcs Williams-Hawkings equation. Simulations have been carried out for various speeds, resulting in a wide range of Reynolds number, which includes subcritical, critical, and supercritical flow states. The results have been compared with experiments and give good agreement. As the pantograph arms are inclined to the flow, the effect of yaw angle is analyzed in this paper and the effect on vortex-shedding frequency and noise level is determined. Moreover, it is demonstrated that the critical Reynolds number, which determines the beginning of the critical flow state, is affected by the yaw angle. In addition, considering the high turbulence intensity on the train roof, a turbulent inflow with different levels of intensity has been considered and the results will be presented.

Session 1pPP

Psychological and Physiological Acoustics: Psychoacoustics (Poster Session)

Nathaniel Spencer, Chair
Communication Sciences and Disorders, University of Pittsburgh, 5056 Forbes Tower, Pittsburgh, PA 15213

Posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to view other posters, contributors of odd-numbered papers will be at their posters from 1:15 p.m. to 2:45 p.m., and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 4:30 p.m. There will be a 15-minute break from 2:45 p.m. to 3:00 p.m.

Contributed Papers

1pPP1. Pitch perception: Spectral and temporal integration at very high frequencies. Bonnie K. Lau and Andrew J. Oxenham (Univ. of Minnesota, 27 East River Rd, N625 Elliot Hall, Minneapolis, MN 55455, bonniekwlau@gmail.com)

Recent work has shown that complex pitch can be extracted from spectrally resolved harmonics, even when all the harmonics are higher than 6 kHz, and so unlikely to be represented by a temporal phase-locked neural code. This study measured spectral and temporal integration of frequency and fundamental-frequency (F0) discrimination at such high frequencies and compared the patterns of results to those obtained at lower frequencies. Difference between the present and earlier studies included the use of level roving on individual components to reduce loudness cues, and the careful control of audibility at very high frequencies. The low and high spectral regions consisted of harmonics 6–10 of two nominal F0s (280 and 1400 Hz). F0 difference limens (DLs) of the complexes and DLs of each harmonic individually were measured at two durations (30 and 210 ms) to measure spectral and temporal integration effects. All tones were presented in background noise to mask distortion products. Preliminary data show similar temporal integration at low and high frequencies, but very different spectral integration, with high-frequency F0DLs generally lower than predicted by a simple detection-theory model. The results suggest different integration mechanisms in the two regions. [Work supported by NIH grant R01DC05216.]

1pPP2. Effects of resolved and unresolved harmonics in the perception of multiple pitches. Jackson E. Graves and Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 East River Parkway, Minneapolis, MN 55455, grave276@umn.edu)

The mechanisms underlying pitch perception have been studied for several decades. Most studies have concentrated on the pitch of single sounds, some have investigated the pitch of two simultaneously presented sounds, but few if any have investigated pitch perception with three or more simultaneous pitches. In contrast, in music it is rule, rather than the exception, that we are presented with multiple pitches. Here, pitch perception was studied with three simultaneously presented harmonic complexes, filtered into a single bandpass region to control the extent to which harmonics from each of the three tones were spectrally resolved. The tones comprised the three possible inversions of the major or minor triad, and the listener's task was to discriminate between major and minor. Preliminary data suggest that subjects who could reliably discriminate major and minor triads in a pure tone training task were also capable of this discrimination when tone complexes were filtered such that the combination of harmonic components should have been unresolved on the cochlea. This finding suggests resolved harmonics may not be necessary to extract the pitch from multiple complex tones. Predictions from various spectral and temporal models of pitch were compared with the results. [Work supported by NIH grant R01DC05216.]

1pPP3. On the search of the optimal range and number of lags for an autocorrelation model of pitch. Sebastian Ruiz-Blais and Arturo Camacho (Universidad de Costa Rica, Guadalupe, Goicoechea, San José 1385-2100, Costa Rica, ruizble@yahoo.com)

Autocorrelation-based pitch models account for a number of pitch phenomena. Previous work has demonstrated their successfulness in response to different stimuli, when combined with a model of the auditory nerve activity and summed across channels [Balaguer-Ballester, Denham & Meddis, J. Acoust. Soc. Am. **124**, 2186–2195 (2008)]. However, autocorrelation models are generally thought of as biologically unrealistic because they require many calculations and might not tally with the neurobiological principle of economy. The present work consists of a series of simulations that aim to determine whether all operations required to compute autocorrelation are strictly necessary to account for the psychophysical data or not. To investigate this, different ranges and densities of lags are used, and the produced implementations are tested with stimuli comprising pure tones, resolved and unresolved complex tones, inharmonic complex tones, and click trains. As a result, the optimal range and number of lags is found, as the most economical setup that agrees with psychophysical evidence.

1pPP4. Relation between measures of psychoacoustic abilities and masking release for unprocessed, envelope, and temporal fine-structure speech in listeners with normal and impaired hearing. Agnes C. Leger (School of Psych., Univ. of Manchester, Ellen Wilkinson Bldg., Oxford Rd., Manchester M13 9PL, United Kingdom, agnes.leger@manchester.ac.uk), Joseph G. Desloge, Charlotte M. Reed, and Louis D. Braida (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Boston, MA)

Léger *et al.* [J. Acoust. Soc. Am. **135**, 2160, 2014] studied masking release in normal-hearing and hearing-impaired listeners using unprocessed speech and speech processed to convey envelope (ENV) or temporal fine-structure (TFS) cues. Consonant identification scores in speech-shaped background noise (continuous and square-wave interrupted at a rate of 10 Hz) indicated a substantial masking release (MR, better performance in the interrupted noise) for all speech types for normal-hearing listeners but only for the TFS speech in most hearing-impaired listeners. The current study was concerned with relating consonant identification scores and masking release to SRTs for sentences in both types of noise as well as to measurements of basic psychoacoustic abilities. These measures included absolute pure-tone detection thresholds, estimates of auditory bandwidth using a notched-noise procedure, and estimates of cochlear compression obtained using a forward-masking paradigm. For both unprocessed and ENV speech, intelligibility scores and MR were correlated with absolute thresholds and auditory bandwidths in the mid-frequency region (but not at a lower frequency). These correlations were less strong for TFS speech intelligibility and did not predict MR. Cochlear compression was not related to speech intelligibility when controlling for the effect of absolute thresholds. [Work supported by NIH R01 DC000117.]

1pPP5. Sound source similarity influences change perception during complex scene perception. Kelly Dickerson (Human Res. and Eng. Directorate, Army Res. Lab., 131 Waldon Rd., Abingdon, MD 21009, dickersonkelly23@gmail.com), Jeremy Gaston, Ashley Foots, and Timothy Mermagen (Human Res. and Eng. Directorate, Army Res. Lab., Aberdeen Proving Ground, MD)

Everyday listening involves identifying an ever-changing milieu of sound sources in the environment. Recent studies have demonstrated that change perception during complex listening tasks is highly error prone; errors can exceed 30% for sounds that are clearly detectable and identifiable in isolation. This change deafness has been generally attributed to failures of attention or memory. The current study investigates the possibility that lower-level informational effects such as acoustic or semantic similarity are predictive of change perception errors. Listeners were briefly presented with pairs of auditory scenes. In experiment 1, the listeners' were asked to identify *if* a change in sound source occurred. In experiment 2, listeners were asked to indicate *where* a change occurred across an array of loudspeakers. Results indicate that performance on the change identification and localization task was strongly influenced by the degree of similarity between the changed source and the "background" sources. Further, the semantic heterogeneity of the background seemed to be important predictor of performance on these change perception task.

1pPP6. Forward masking in children and adults with normal hearing and hearing loss. Marc A. Brennan, Walt Jesteadt, and Ryan McCreery (Res., Boys Town National Res. Hospital, 555 N. 30th St., Omaha, NE 68131, marc.brennan@boystown.org)

The objective was to determine the extent to which childhood hearing loss can affect the development of temporal resolution. Participants were divided into those with and without hearing loss. Participants included 16 children per group (normal hearing, hearing loss) and 14 adults per group. Forward-masked thresholds for a 2 kHz masker and target (10 ms signal delay) were measured at similar points on the dynamic range for the participants with normal hearing and hearing loss. As expected, the amount of masking decreased with age. Due to their hearing loss, the participants with hearing loss were tested at lower masker sensation levels than the participants with normal hearing. For that reason, it was expected that less masking would occur for the participants with hearing loss, and this was observed in the data. It was hypothesized that age and hearing loss would interact, such that the younger children with hearing loss would show greater masking than younger children with normal hearing. Instead, both children with normal hearing and with hearing loss showed higher masking than their adult counterparts, suggesting that the amount of masking did not interact with age and hearing status. These findings suggest that children show less efficient temporal processing.

1pPP7. Dynamic binaural sound source localization with interaural time difference cues: Artificial listeners. Liang Sun (Dept. of Elec. and Comput. Eng., The Univ. of Texas at San Antonio, San Antonio, TX), Xuan Zhong, and William Yost (Arizona State Univ., 975 S Myrtle Ave., Lattie F Coor Hall 2211, Tempe, AZ 85287, xuan.zhong@asu.edu)

When an array of two acoustic sensors is used to localize sound sources based on time differences alone, possible solutions form a cone of confusion. This study, together with a similar one for human listeners, demonstrates that azimuth/vertical planes localization of a sound source using only time difference information is feasible when self-motion measurement of the listener is available. In particular, the case of a static sound source playing that broadcasts low frequency pure tone signals was investigated. A dummy head is mounted on top of a rotating chair to mimic the head and body motion of human beings, as well as to collect audio signals. A gyroscope was mounted on top of the dummy head to collect self-motion data. A mathematical model was constructed to describe the interaural time difference (ITD) change over time, and an Extended Kalman Filter (EKF) was used to estimate the spatial angles of the sound sources with respect to the listener using the developed mathematical model and measured data. The effectiveness and robustness of the developed algorithm are shown by both the numerical and experimental results, which reveal the quick convergence of the estimated spatial angles toward their real values given noisy measured data. The possibilities of using other spatial hearing cues were also discussed.

1pPP8. On the extent of head motion in listening tests and individual measurements using different head-tracking sensors. György Wersényi (Dept. of Telecommunications, Széchenyi István Univ., Egyetem tér 1, Györ 9026, Hungary, wersenyi@sze.hu) and Jeff Wilson (Interactive Media Technol. Ctr., Georgia Inst. of Technol., Atlanta, GA)

Measurements and applications in spatial hearing research, virtual auditory displays, etc., often rely on head-related transfer functions (HRTFs) of human subjects. Individually measured HRTFs have the advantage of being more accurate in virtual localization tasks. On the other hand, the measurement and recording procedure raise several new problems such as signal-to-noise ratio issues and subject comfort. For measurements with human subjects, lots of methods are used from free heads to different head fixations methods. This study analyses the extent of head movement during measurements using different sensors and environmental conditions based on the circular angle variance, errors in yaw-pitch-roll directions and magnitude of standard deviation. Conclusive results indicate magnitudes of standard deviation of 2–8 cm and errors about 2 degrees depending on the situation as well as a preference for sitting instead of standing posture.

1pPP9. Effects of harmonic roving on pitch discrimination. Sébastien Santurette, Mathilde Le Gal de Kérangal, and Suyash N. Joshi (Hearing Systems, Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, DTU Bygning 352, Kgs. Lyngby 2800, Denmark, ses@elektro.dtu.dk)

Performance in pitch discrimination tasks is limited by variability intrinsic to listeners which may arise from peripheral auditory coding limitations or more central noise sources. The present study aimed at quantifying such "internal noise" by estimating the amount of harmonic roving required to impair pitch discrimination performance. Fundamental-frequency difference limens (F0DLs) were obtained in normal-hearing listeners with and without musical training for complex tones filtered between 1.5 and 3.5 kHz with F0s of 300 Hz (resolved harmonics) and 75 Hz (unresolved harmonics). The harmonicity of the tone complexes was varied by systematically roving the frequency of individual harmonics, which was taken from a Gaussian distribution centered on the nominal frequency in every stimulus presentation. The amount of roving was determined by the standard deviation of this distribution, which varied between 0% and 16% of the tested F0. F0DLs for resolved harmonics remained unaffected for up to 6% roving, and increased thereafter. For unresolved harmonics, performance remained stable up to larger roving values. The results demonstrate a systematic relationship between F0DLs and stimulus variability that could be used to quantify the internal noise and provide strong constraints for physiologically inspired models of pitch perception.

1pPP10. Pitch perception and frequency-following responses elicited by lexical-tone chimeras in American and Chinese adults. Fuh-Cherng Jeng (Commun. Sci. and Disord., Ohio Univ., 1 Ohio University Dr., Grover Ctr. W224, Athens, OH 45701, jeng@ohio.edu), Meng-Shih Chou, Chia-Der Lin (China Medical Univ. Hospital, Taichung City, Taiwan), John Sabol, Grant Hollister (Commun. Sci. and Disord., Ohio Univ., Athens, OH), Ching-Hua Chen (China Medical Univ. Hospital, Taichung City, Taiwan), Jessica Kenny (Commun. Sci. and Disord., Ohio Univ., Athens, OH), and Yung-An Tsou (China Medical Univ. Hospital, Taichung City, Taiwan)

Objective: Previous research has shown the usefulness of utilizing auditory chimeras (i.e., interchanging the envelope and fine structure of two different sounds) in assessing a listener's perception of the envelope and fine structure for an acoustic stimulus. However, research comparing and contrasting behavioral and electrophysiological responses to this stimulus type is scarce. *Design*: Two sets of chimeric stimuli were constructed by interchanging the envelopes and fine-structures of the rising /yi^2/ and falling /yi^4/ Mandarin pitch contours that were filtered through 1, 2, 4, 8, 16, 32, and 64 frequency banks. Behavioral pitch-perception tasks (through a single-interval, two-alternative, forced-choice paradigm) and electrophysiological responses (through the frequency-following response to voice pitch using validated scalp-recorded potential methods) were obtained from two groups of participants (native speakers of a tonal or nontonal language). *Study Sample*: Thirteen American and 13 Chinese adults were recruited. *Results*: A two-way analysis of variance showed significance ($p<0.05$) within and across the filter bank and language background factors for the behavioral measurements to the lexical-tone chimeras, while the frequency-following response demonstrated a significance across filter banks, but not for language background. *Conclusions*: Frequency-following responses to voice pitch provide supplementary information on how chimeric stimuli are processed at the brainstem level.

1pPP11. Effects of compression channel number and linked compression on masked lateralization performance in simulated bilateral cochlear implant listening. Nathaniel Spencer and Christopher Brown (Commun. Sci. and Disord., Univ. of Pittsburgh, 5056 Forbes Tower, Pittsburgh, PA 15213, njs64@pitt.edu)

Bilateral cochlear implants (CIs) are becoming increasingly common. It is hoped for bilateral CI users that audibility in quiet will be complemented by performance benefits in everyday binaural listening tasks, like location-identification of a speech talker presented amid other talkers. While normal-hearing (NH) listeners have typically performed well in such kinds of tasks, bilateral CI users have performed poorly. It is possible that, for bilateral CI users, better performances can be attained through a re-thinking of device signal-processing strategies. In particular, we are interested in evaluating alternatives to single-channel independent compression, as is typically applied at the automatic gain control (AGC) stage. Specifically, we test the hypothesis that for simulated bilateral CI users (listeners with both NH and hearing impairments, HI), lower rms-error in a masked lateralization task will be achieved for multi-channel AGC and linked AGC, than for single-channel independent AGC, with and without pre-emphasis filtering, and for low and high compression thresholds. Preliminary data suggest a lower rms-error for linked compression than for independent compression, in conjunction with no detrimental effect of multi-channel compression. These data support the possibility of linked AGC with multi-channel AGC as an alternative to independent AGC in bilateral CI listening.

1pPP12. The build-up and resetting of auditory stream segregation: Effects of timbre, level, and abrupt change. Saima L. Rajasingam, Robert J. Summers, and Brian Roberts (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, rajasisl@aston.ac.uk)

Two experiments explored the effects of timbre, level, and abrupt change on the dynamics of streaming. Listeners continuously monitored 20-s ABA-sequences (1-kHz base frequency; $\Delta f = 4–8$ semitones) and reported when the sequence was heard as integrated or segregated. Experiment 1 used pure tones and narrowly spaced (± 25 Hz) tone pairs (dyads); both evoke similar excitation patterns but dyads have a "rougher" timbre. Dyad-only sequences induced a strongly segregated percept, with limited scope for further build-up. Abrupt alternations every 13 triplets produced large changes in the extent of segregation. Dyad-to-pure transitions produced substantial resetting, but pure-to-dyad transitions elicited greater segregation than for the corresponding time in dyad-only sequences (overshoot). Experiment 2 examined the effect of varying triplet level. Pure-tone sequences with a maintained 12-dB difference showed similar build-up, but an asymmetry occurred for alternating sequences. Increased-level transitions caused a significant reduction in segregation (resetting) whereas decreased-level transitions had little or no effect. The results suggest that timbre can strongly affect the likelihood of stream segregation, even without peripheral-channeling cues, but moderate level differences do not. Furthermore, abrupt changes in sequence properties have variable effects on the tendency to report segregation—including resetting, overshoot, and asymmetries in the effect of transition direction.

1pPP13. Individual differences in acoustic acuity at supra-threshold level. Lengshi Dai and Barbara G. Shinn-Cunningham (Dept. of Biomedical Eng., Boston Univ., 677 Beacon St., Boston, MA 02215, ldai@bu.edu)

The ability to detect acoustic amplitude modulated envelopes differs even among listeners with normal hearing thresholds (NHTs), an effect that to some degree may correlate with noise exposure history. Moreover, at least one animal study suggests that auditory nerve fibers (ANFs) with lower spontaneous firing rates (high thresholds) are more vulnerable to damage from noise exposure compared to those with high spontaneous rate ANFs. Given that NHTs depend primarily on high-spontaneous rate fibers, even relatively severe loss of low-spontaneous ANFs could be present in listeners that have "normal hearing" according to traditional hearing screenings. Here, we evaluated the robustness of supra-threshold coding, which may relate to low-spontaneous ANFs health, in listeners with NHTs. Specifically, we measured the phase-locking-values (PLVs) of subcortical envelope following responses to SAM tones (carrier: 1500 Hz; modulation rate: 100 Hz) at two sensation levels (20/50 dB SL) and with two modulation depths (0/-6 dB). Results reveal huge individual differences in PLV strengths that are negatively correlated with behavioral amplitude modulation detection thresholds for noises at a moderate supra-threshold level (carrier: band-pass-filtered noise centered at 1500 Hz with 300 Hz bandwidth; modulation rate:10 Hz). Our data support the hypothesis that the number of operational low-spontaneous rate ANFs may differ human listeners with NHTs, and that deficit in low-spontaneous rate ANFs can lead to deficits in perceptual abilities.

1pPP14. Effects of active and passive hearing protective devices on sound source localization, tone detection, and speech recognition. Andrew D. Brown, Brianne T. Beemer, Nathaniel T. Greene, and Daniel J. Tollin (Physiol. & Biophys., Univ. of Colorado School of Medicine, 12800 East 19th Ave., RC1-N, Rm. 7401G, M.S. 8307, Aurora, CO, Andrew.D.Brown@ucdenver.edu)

Hearing protective devices (HPDs) such as earplugs and earmuffs offer to mitigate noise exposure and thus noise-induced hearing loss among persons frequently exposed to intense sound, e.g., military personnel or industrial workers. However, distortions of spatial acoustic information and attenuation of low-intensity sounds caused by many existing HPDs can make their use untenable in high-risk environments where auditory situational awareness is imperative. Here, we assessed (1) sound source localization accuracy using a head-turning paradigm, (2) tone detection thresholds using a two-alternative forced-choice task, and (3) speech-in-noise recognition using a modified version of the QuickSIN test in 10 young normal-hearing males wearing four different HPDs (two active, two passive), including two new and previously untested devices. Relative to unoccluded (control) performance, all tested HPDs significantly degraded performance across tasks, although one active HPD slightly improved high-frequency tone detection thresholds, and did not degrade speech recognition. Behavioral data were examined with respect to binaural acoustic information (directional transfer functions) measured in a binaural manikin with and without tested HPDs. Data reinforce previous reports that HPDs significantly compromise auditory perceptual facilities, particularly sound localization due to distortions of high-frequency pinna cues.

1pPP15. Sequential streaming under reverberation: Effects of the reverberant tail properties. Eugene J. Brandewie and Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, 75 E. River Rd., Minneapolis, MN 55455, ebrandew@umn.edu)

A physical difference between two alternating stimuli can elicit perceptual segregation when the difference is sufficiently salient. One such cue involves differences in reverberation, potentially caused by differences in source distance. Here, we studied what aspects of difference in reverberation are most important in eliciting segregation using a rhythmic masking task. Two interleaved sequences of Gaussian noise bursts (target and interferer) were presented on each trial and listeners attempted to identify which of two rhythms was presented in the target sequence. The influence of the reverberation tail (or damped decay) was studied by parametrically changing its duration in the target sequence, while eliminating all binaural cues. The influence of spectral content of the tail was examined by simulating the spectral coloration produced by real rooms. Results suggest that damped tails can elicit perceptual segregation with tail durations less than 100 ms. In addition, the spectral content of the tail can further influence segregation performance. Overall, differences in reverberation can serve as a prominent cue to aid perceptual segregation, particularly if a room environment introduces differences in spectrum based on distance, which is often the case in real rooms. [Work supported by NIH grant R01DC07657.]

1pPP16. Streaming and sound localization with a preceding distractor. Gabriela Andrejkova (Inst. of Comput. Sci., P. J. Safarik Univ. in Kosice, Jesenna 5, Kosice 04001, Slovakia, gabriela.andrejkova@upjs.sk), Virginia Best, Barbara G. Shinn-Cunningham (Boston Univ., Boston, MA), and Norbert Kopčo (Inst. of Comput. Sci., P. J. Safarik Univ. in Kosice, Kosice, Slovakia)

A previous study of sound localization with a preceding distractor showed that (1) the distractor affects response bias and response variance for distractor-target inter-stimulus-intervals of up to 400 ms, and that (2) localization responses are biased away from the distractor even on interleaved control trials in which the target is presented alone [Kopco *et al*., JASA **121**, 420–432, 2007]. Neural mechanisms operating on time scales of milliseconds to tens of seconds are likely to cause to these effects. The current study examined how perceptual organization affects target localization performance. Sound localization was examined for 2-ms click target stimuli. On 80% of trials, the target was preceded by a distractor, designed either to be grouped with the target (distractor was an identical 2-ms click) or to be perceived in a separate stream (an isochronous train of eight clicks whose inter-click-interval was different from the distractor-target inter-stimulus-interval). As hypothesized, the single-click distractor affected target localization more than the eight-click distractor. On the other hand, the biases in the control trials were greater for the eight-click distractor. These results indicate that performance is influenced by both top-down mechanisms like streaming and bottom-up mechanisms like stimulus distribution-based adaptation. [Work supported by APVV-0452-12 and R01DC009477.]

1pPP17. Multiple sound source localization when sounds are stationary or rotating. Xuan Zhong and William Yost (Arizona State Univ., 975 S Myrtle Ave., Lattie F Coor Hall 2211, Tempe, AZ 85287, xuan.zhong@asu.edu)

In the field of spatial hearing, it is known that the sound source localization error increases when distractors are present. This study first determined the maximum number of sound sources that can be perceived. In one experiment, listeners determined how many sources (loudspeakers) presented sounds and these loudspeaker locations. In another experiment, listeners were asked to locate a new sound source which was added to an existing set of fixed sources. Listeners determined up to about four sources producing simultaneous unrelated speech sounds in both experiments. A second part of the research examined the effect of rotating the sound sources on the listener's perception. If multiple unrelated sounds such as speech rotated, listeners could much more easily determine sound rotation than if the sounds were related to each other. If the sounds were related (e.g., all sounds were harmonics of a fundamental) but were still presented from a number of different loudspeakers, listeners could hardly process the individual sounds even when sounds rotated among the loudspeakers. Thus, the ability to locate multiple sound sources depends on the total number and motion of sources and the type of sound. [Research supported by an AFOSR grant.]

1pPP18. The addition of a second lag to the lead-lag precedence effect paradigm for temporally overlapping noise stimuli. M. Torben Pastore and Jonas Braasch (Ctr. for Cognition, Commun., and Culture, Rensselaer Polytechnic Inst., 4 Irving Pl., Troy, NY 12180, m.torben.pastore@gmail.com)

In reverberant conditions, humans routinely perceive sound sources in the direction of the first wavefront despite competing directional information presented by a host of reflections arriving soon after—the so-called "Precedence Effect." This is often tested over headphones using a "direct sound" (lead) with a single delayed copy (lag) serving as a modeled reflection. Previously, we employed this common experimental paradigm to investigate the lateral extent of the precedence effect using temporally overlapping noise stimuli. The current study extends this inquiry towards the multiple reflections encountered in room-acoustic scenarios by presenting a second lag. Lead and lag stimuli are 200-ms Gaussian noise (500-Hz center frequency, 800-Hz bandwidth) presented dichotically with a programmable amount of delay for both lags. Relative to the intensity of the lead, the two lags are presented at 0, -3, and -6 dB. The lead is presented at the midline, with an ITD of 0 μs. The two lags are delayed by between 1 and 7 ms, with opposing ITDs of ± 300 μs. Listeners indicate the lateralization of their auditory event with an acoustic pointer.

1pPP19. Relation between temporal envelope coding, pitch discrimination, and compression estimates in listeners with sensorineural hearing loss. Federica Bianchi, Sébastien Santurette, Michal Fereczkowski, and Torsten Dau (Tech. Univ. of Denmark, Ørsteds Plads Bldg. 352, Lyngby, Denmark 2800, Denmark, fbia@elektro.dtu.dk)

Recent physiological studies in animals showed that noise-induced sensorineural hearing loss (SNHL) increased the amplitude of envelope coding in single auditory-nerve fibers. The present study investigated whether SNHL in human listeners was associated with enhanced temporal envelope coding, whether this enhancement affected pitch discrimination performance, and whether loss of compression following SNHL was a potential factor in envelope coding enhancement. Envelope processing was assessed in normal-hearing (NH) and hearing-impaired (HI) listeners in a behavioral amplitude-modulation detection task. Fundamental frequency difference limens (F_0DLs) were obtained in the same listeners for complex tones with varying harmonic resolvability. Basilar-membrane input/output functions were measured to assess individual compression ratios. For NH listeners, F_0DLs decreased with increasing harmonic resolvability. For the unresolved conditions, all five HI listeners performed as good as or better than NH listeners with matching musical experience. Two HI listeners showed lower amplitude-modulation detection thresholds than NH listeners for low modulation rates, and one of these listeners also showed a loss of cochlear compression. Overall, these findings suggest that some HI listeners may benefit from an enhancement of temporal envelope coding in pitch discrimination of unresolved complex tones, and that this enhancement may be also ascribed to a reduction of cochlear compression following SNHL.

1pPP20. Time dependence of the time interval in distance/ intensity threshold in human listeners. Larisa Dunai, Ismael Lengua, and Miguel Iglesias (Universitat Politècnica de València, St. Camino de Vera s/n 5L, Valencia 46022, Spain, ladu@upv.es)

The paper presents the distance threshold for complex virtual sounds localization in dependence of the time interval. The method of measurement was a forced-choice, three-down one-up staircase. The sound is a complex record of a click sound of 47 ms at 44.1 kHz measured by using a maximum length binary sequence in an anechoic chamber. The azimuth of the experimental is 0°, at the center of the human head where the ITD = 0, varying just the intensity. The investigation reported that the lowest distance thresholds of 0.06 m occurred for high inter click intervals at 200 ms–300ms. At lower ICIs, the distance threshold increases considerably. The experimental was carried out with the objective to find the distance threshold and time interval for implementing the data for a navigation device for blind people.

1pPP21. Time-efficient multidimensional threshold tracking method. Michal Fereczkowski, Borys Kowalewski, Torsten Dau, and Ewen N. MacDonald (Elektro, DTU, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, mfer@elektro.dtu.dk)

Traditionally, adaptive methods have been used to reduce the time it takes to estimate psychoacoustic thresholds. However, even with adaptive methods, there are many cases where the testing time is too long to be clinically feasible, particularly when estimating thresholds as a function of another parameter, such as in temporal masking curves or when characterizing auditory filters. Here we present a new method, the "grid" method, which adaptively varies multiple parameters during each experimental run. By changing the way the parameter-response space is sampled, the method increases the proportion of experimental time spent in the vicinity of the sought-after threshold curve. The resulting increase in time-efficiency is substantial and can make some measurements clinically feasible. Thresholds from temporal masking curves obtained with the grid method are compared with those from one of the most time-efficient standard methods (single-interval up-down adaptive method of Lecluyse, 2013). Overall, individuals' results from both methods are very highly correlated, but the grid method was an order of magnitude faster in estimating thresholds. The application of the grid method to other measurements, such as characterizing auditory filters, will also be discussed.

1pPP22. Measuring rapid adaptation to complex acoustic environments in normal and hearing-impaired listeners. Sofie Aspeslagh (MRC/CSO Inst. of Hearing Res.—Scottish Section, Glasgow Royal Infirmary, 16 Alexandra Parade, Glasgow G31 2ER, United Kingdom, sofie@ihr.gla.ac.uk), Fraser Clark (School of Eng. and Computing, Univ. of the West of Scotland, Paisley, United Kingdom), Michael A. Akeroyd, and W. O. Brimijoin (MRC/CSO Inst. of Hearing Res. – Scottish Section, Glasgow, United Kingdom)

Listeners can move from room to room while conversing, encountering changes in the acoustics of their surroundings. In order to maintain speech intelligibility when faced with changing noises and reverberation listeners may rapidly adapt by building and updating a model of their surrounding auditory space. We have previously demonstrated rapid adaptation to a change in the acoustic environment, consisting of a brief decrease and then a recovery in intelligibility. In the current study we investigated whether the amount and time-course of this adaptation changes as a function of age and/or hearing impairment, and whether the amount of reverberation or the complexity of the background noise would affect adaptation time. We asked 29 normal-hearing and 37 hearing-impaired listeners to identify an ongoing stream of random target words, presented at a rate of one every 1.5 seconds, while the acoustic environment was switched every nine seconds. On average, intelligibility dropped by 16% upon entering a new environment before recovering to ceiling performance within 2.3 seconds. These results were not affected by hearing impairment or age. Preliminary analyses suggest that listeners may require additional time to adapt noises of greater complexity. [Work supported by MRC (U135097131) and the Chief Scientist Office (Scotland).]

1pPP23. An adaptive procedure for estimating the auditory-filter phase response. Niall A. Klyn and Lawrence L. Feth (Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd, 110 Pressey Hall, Columbus, OH 43210, klyn.1@osu.edu)

Current techniques for psychophysical estimates of the phase curvature of the basilar membrane rely on laborious data collection. For each point on the basilar membrane a separate threshold estimate — more commonly several estimates — must be found for each phase relationship. This practically limits the precision of the estimates and the number of subjects for whom estimates may be obtained. Here, we propose a rudimentary adaptive technique for rapidly estimating the phase curvature of the basilar membrane psychophysically. Consistent with the extant data the procedure assumes that the underlying threshold versus masker curvature function has a U shape. Rather than find thresholds at each value of masker curvature, the procedure seeks out only the value which produces the least masking. It does so by dropping the signal level until only three adjacent masker curvature values fail to mask the signal. New upper and lower boundaries for the masker curvature are then set, the step size between masker curvature values is reduced, and the previous steps are repeated. We present a comparison of the estimates produced by the conventional techniques and the proposed adaptive technique, as well as their relative efficiencies.

1pPP24. Measuring the auditory phase response based on interaural time differences. Hisaaki Tabuchi, Bernhard Laback, Piotr Majdak, Thibaud Necciari, and Katherina Zenke (Austrian Acad. of Sci., Acoust. Res. Inst., Wohllebengasse 12-14, Vienna 1040, Austria, tabuchi@kfs.oeaw.ac.at)

Harmonic complexes are often used as maskers for measuring the cochlear phase response. The phase curvature can affect the masking in order of up to 20 dB, an effect known as the masker-phase effect. There is evidence that signals yielding peaky internal masker representations after passing the cochlear filter produce minimum masking, with the fast-acting cochlear compression as the main contributor to that effect. Thus, in hearing-impaired listeners showing reduced or absent compression, the estimation of phase response using the masking method may be difficult. Here, an alternative method is proposed, which is based on the effect of signal peakedness on the sensitivity to interaural time differences (ITD) in the signal envelope. With the two methods, ITD and masking thresholds were measured, respectively, in seven normal-hearing listeners. The stimuli were 300-ms Schroeder-phase harmonic complexes, ranging from 3400 to 4600 Hz with a 100-Hz fundamental frequency, with the signal phase curvature varied between -1 and 1. The lowest ITD thresholds were observed for phase curvatures that produced minimum masking. This suggests that the ITD method is a promising candidate for estimating the cochlear phase response in hearing-impaired listeners.

1pPP25. Cochlear fine structure predicts behavioral decision weights in a multitone level discrimination task. Jungmee M. Lee (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, 1410 E. Skyline Dr., Madison, WI 53705), Glenis Long (Speech, Lang. and Hearing Sci., Graduate Ctr., City Univ. of New York, New York, NY), Inseok Heo, Christophe Stoelinga, and Robert Lutfi (Commun. Sci. and Disord., Univ. of Wisconsin–Madison, Madison, WI, ralutfi@wisc.edu)

Listeners show highly replicable, idiosyncratic patterns of decision weights across frequency affecting their performance in multi-tone level discrimination tasks. The different patterns are attributed to peculiarities in how listeners attend to sounds. However, evidence is presented in the current study that they reflect individual differences in cochlear micromechanics, which can be evaluated using otoacoustic emissions (OAEs). Spontaneous OAEs (SOAEs) and the fine structure of stimulus-frequency OAEs (SFOAEs) were measured in a group of normal-hearing listeners. The same group of listeners performed a two-tone, sample-level discrimination task wherein the frequency of one tone was selected to correspond to a SOAE and the other was selected well away from a SOAE. Tone levels were either 50 or 30 dB SPL. The relative decision weight of the two tones for each listener and condition was estimated from a standard COSS analysis of the trial-by-trial data [Berg (1989), J. Acoust. Soc. Am. **86**, 1743–1746]. A strong linear relation was observed between the average relative decision weight and the average relative level of both SOAE and SFOAE.

1pPP26. A meta-analysis of the effects of hearing impairment and hearing aids on directional hearing. Michael Akeroyd and William M. Whitmer (MRC/CSO Inst. of Hearing Res. - Scottish Section, New Lister Bldg., Glasgow Royal Infirmary, Glasgow, Strathclyde G31 2ER, United Kingdom, maa@ihr.gla.ac.uk)

We report here meta-analyses of all the experiments that have been done since the 1980s on how accurate hearing-impaired listeners are at determining the spatial direction of sound sources. The results demonstrate that their performance is somewhat worse than normal hearing listeners for all directions, and especially so for sounds presented from the side (including distinguishing front vs. back) or for changes in elevation. There is considerable variation across listeners and experiments. In general, hearing aids do not improve performance, and there is overall little effect of differences in hearing aid features or designs (e.g., for left/right accuracy, an across-experiment mean difference of just 1 degree of RMS error). In many situations, a unilateral fitting results in a localization deficit. Though statistically significant effects of aided localization have been observed experimentally, few of them are generalizable as they often occurred for just some source directions, stimuli, or groups of listeners. Overall, there is no experimental evidence for a substantial, universal benefit from hearing aids to directional accuracy.

1pPP27. Effect of spectrally-remote maskers on sentence recognition by adults and children. Carla L. Youngdahl, Sarah E. Yoho, Rachael Frush Holt, Frederic Apoux, and Eric W. Healy (Speech & Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, carla.youngdahl@gmail.com)

Adults display improved detection of a signal in noise when the spectral frequency of that signal is known, relative to when it is unknown. In contrast, infants do not display this improvement, suggesting that they monitor all frequencies equally, even when it is not advantageous to do so. To assess the impact of this "spectral attention" development during speech recognition, sentences in noise were lowpass filtered at 1500 Hz and presented along with spectrally remote low-noise noise maskers that produced no spectral overlap of peripheral excitation. As anticipated, sentence recognition by adults was not affected by the presence of remote maskers, irrespective of their bandwidth and spectral location. This result was also observed in a group of 7-year-old children. However, the youngest children tested (5-year-olds) displayed poorer sentence recognition in the presence of the remote maskers, suggesting that they were unable to focus attention on the spectral region of speech. The current results suggest that an important step occurs in the development of auditory spectral attention around age 6. These results may also help explain why children typically require more favorable signal-to-noise ratios than adults to achieve similar levels of speech performance in noise. [Work supported by NIH.]

1pPP28. Adult listener's detection thresholds for vowels and two-syllable words in different two-talker maskers. Monika-Maria Oster and Lynne A. Werner (Speech and Hearing Sci., Univ. of Washington, 1417 North East 42nd St., Seattle, WA 98105, mmoster@uw.edu)

There is an increased interest in using natural speech targets and speech maskers to assess infants' auditory skill development. While providing ecological validity, natural speech is an inherently uncontrolled signal. Discrepancies between existing studies may be the result of differences in the maskers and/or target stimuli used. This study explores adult listener's detection thresholds for vowels and two-syllable words in three different two-talker maskers to assess the impact of (1) masker characteristics and (2) differences in target signals. Each two-talker masker was created by recording two female speakers reading different passages in monotone, in adult-directed style, or infant-directed style. Silences longer than 300 ms were deleted, and the passages' root-mean-square amplitudes balanced before combining them. The target stimuli consisted of the vowels /a/, /o/, /e/ and the words "baby" and "ice-cream." They were spoken by different female talkers in both adult-directed and infant-directed styles. An adaptive two-interval forced choice method was used to estimate threshold. Preliminary results suggest that there are small differences in detection thresholds between the different two-talker maskers. Larger differences are found between the thresholds of different target signals, indicating that discrepancies between existing studies may be due to differences in target stimuli used.

1pPP29. Effect of listener experience on pitch and rhythm perception with tonal sequences. Sandra J. Guzman, Cody Elston (Audio Arts & Acoust., Columbia College Chicago, 33 East Congress Parkway, Ste. 600, Chicago, IL 60605, sguzman@colum.edu), Valeriy Shafiro, and Stanley Sheft (Communications Disord. & Sci., Rush Univ., Chicago, IL)

Experience and training can influence discrimination of tonal sequences. Current work investigated pitch and rhythm processing of four-tone sequences by audiology and speech students, audio-arts students experienced in critical listening, and trained musicians. Sequence tones either had a fixed duration (212 ms) with frequency randomly selected from a logarithmically scaled distribution (400–1750 Hz), a fixed frequency (837 Hz) with a randomly selected log scaled duration (75–600 ms), or a random frequency and duration. In initial conditions, the task was to assemble sequence elements to recreate the target sequence for each of the three sequence types. To evaluate effect of extraneous randomization, both frequency and duration were randomized in the final two conditions with only one of the two attributes defining the target sequence. Audio-arts students performed significantly better than audiology and speech students in the reconstruction task. In conditions involving joint processing of sequence pitch contour and rhythm, the performance of audio-arts students was well approximated by the optimal combination of uncorrelated but integral stimulus dimensions. Ongoing work is evaluating the performance of trained musicians in the sequence-reconstruction task, with emphasis on manner in which information is combined across the dimensions of sequence pitch and rhythm. [Work supported by NIH.]

1pPP30. Fast frequency selectivity measures in listeners with severe hearing loss. Eric C. Hoover (Univ. of South Florida, 3802 Spectrum Blvd., Ste. 210A, Tampa, FL 33612, erichoover@usf.edu), Michael C. Blackburn (Captain James A. Lovell Federal Health Care Ctr., North Chicago, IL), and Pamela E. Souza (Northwestern Univ., Evanston, IL)

Communication difficulties in severe hearing loss are a combination of elevated pure-tone thresholds and suprathreshold deficits; for example, frequency selectivity, which may be substantially impaired in listeners with severe loss. The purpose of this study was to investigate frequency selectivity measures in listeners with severe loss (pure-tone average 60–90 dB HL). Two tests were used: fast psychophysical tuning curves (PTC), in which an intermittent tone was detected during a narrowband noise masker swept continuously in frequency; and spectral ripple (SR), in which the number of sinusoidal ripples in the spectrum of a broadband noise was tracked in an adaptive forced-choice procedure. Results showed that both tests have advantages when evaluating listeners with severe loss. Fast PTC provided a direct representation of tuning in a specific frequency region but was limited by the residual dynamic range of the listener. Dynamic range limitations were partially overcome by combining data from multiple trials. SR was able to be performed by listeners with minimal dynamic range but the interpretation of results was limited by potential differences in hearing across the bandwidth of the stimuli. Both fast PTC and SR revealed information about suprathreshold deficits that could help guide audiological intervention. [Work supported by NIH.]

1pPP31. Virtual auditory display validation using transaural techniques. Pavel Zahorik (Heuser Hearing Inst. and Div. of Communicative Disord., Dept. of Surgery, Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu) and Zijiang J. He (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

Validation of headphone-based virtual auditory display technology is inherently difficult, given the potential for display hardware to interfere with normal sound field listening conditions due to occlusion of the pinna. This difficulty is eliminated if loudspeaker-based transaural techniques are instead used for the virtual auditory display. Transaural techniques also offer the advantage of being able to use the display loudspeakers, or other additional loudspeakers, as individual reference locations in order to validate the display under natural listening conditions in the same sound field. Such validation can also include explicit comparison to visual localization of the sound sources. In this way, the accuracy and precision of virtual source localization can be directly compared to real source localization, and optionally to visual localization. Results of a validation experiment of this type using non-individualized head-related transfer functions are reported and compared to analogous data from a headphone-based virtual auditory display. [Work supported by NEI.]

1pPP32. Application of a monaural glimpsing model to binaural speech mixtures. Virginia Best, Christine R. Mason, Jayaganesh Swaminathan, Elin Roverud, and Gerald Kidd (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, cmason@bu.edu)

Under certain circumstances, listeners with sensorineural hearing loss demonstrate poorer speech intelligibility in spatially separated speech maskers than those with normal hearing. One important issue in interpreting these results is whether the target speech information is available or audible in the spatialized mixture. Simple energy-based "glimpsing" models can be used to quantify the available target in speech mixtures. For example, as target level decreases in a fixed-level masker, fewer and fewer good glimpses are available. Moreover, those glimpses may not all be above threshold, particularly for individuals with hearing loss. In this study we used a glimpsing model to isolate available target glimpses in binaural speech stimuli (separately for each ear) as a function of target-to-masker ratio. Performance for stimuli processed in this way was compared to binaural performance in listeners with normal hearing and listeners with hearing loss. Results suggest that binaural performance was partly limited by the ability to use the available glimpses. Increasing the level of the glimpses gave mixed results, which will be discussed in terms of audibility. [Work supported by NIH/NIDCD.]

1pPP33. Effects of hearing impairment on sensitivity to dynamic spectral change. Michelle R. Molis, Nirmal Srinivasan, and Frederick J. Gallun (VA RR&D National Ctr. for Rehabilitative Auditory Res., VA Portland Health Care System, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, michelle.molis@va.gov)

The loss of peripheral auditory sensitivity, precise temporal processing, and frequency selectivity associated with hearing loss suggests that the results obtained for pure tone glide stimuli will not necessarily correspond to results obtained with more complex dynamic stimuli for listeners with hearing impairment. Normally hearing (NH) and hearing-impaired (HI) listeners identified changes in frequency as rising or falling both for tone glides and for spectrotemporal ripples. Tones glided linearly up or down in frequency with an extent of 1, 0.66, or 0.33 octaves centered around 500 or 1500 Hz. Ripple stimuli, presented in octave bands centered around 500 or 1500 Hz or in a broadband condition extending from 20–20,000 Hz, had a spectral density of 2 cycles/octave and temporal modulation gliding up or down at rates of 1, 4, or 16 Hz. Sensitivity to dynamic changes was assessed as percent correct direction identification and bias was characterized as the ratio of correctly-identified rising versus falling glides. Substantial individual variability was observed for both measures for both the NH and HI listeners, and the ability to perform the tone glide task was not a consistent predictor of the ability to perform the spectrotemporal ripple identification. [Work supported by NIH/NIDCD.]

1pPP34. Duration and transition effects in multiple-burst, multitone masking. Eric R. Thompson (Ball Aerosp. & Technologies Corp., 2610 7th St, Bldg. 441, Wright Patterson, OH 45433, eric.thompson.ctr@wpafb.af.mil), Matthew G. Wisniewski, Nandini Iyer, and Brian D. Simpson (Air Force Res. Lab, Wright-Patterson AFB, OH)

Previous studies of tone detection using a multiple-burst same (MBS) or multiple-burst different (MBD) masker have shown that tone detection thresholds were lower with an MBD masker than with an MBS masker, and decreased as the number of bursts increased. In those studies, as the number of bursts increased, the total signal duration increased as well as the number of times the (MBD) masker changed frequencies. The present study was designed to differentiate between the effects of signal duration and masker transitions. In one condition, the burst duration was inversely varied with the number of bursts so that the overall stimulus duration was always 480 ms. In a second condition, the stimuli always had eight 60-ms bursts, but the number of times the masker transitioned from one random draw of frequencies to a different draw was varied from zero (MBS) to seven (MBD). The MBD condition showed a steady improvement with number of bursts, regardless of the burst duration. There was a similar improvement with the number of masker transitions. There was only a small improvement in thresholds with increasing burst duration. Results suggest that the number of changes of masker frequencies drives performance more than the total signal duration.

1pPP35. Using multidimensional scaling techniques to quantify binaural squelch. Gregory M. Ellis (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY 40292, g.ellis@louisville.edu), Pavel Zahorik (Heuser Hearing Inst. and Div. of Communicative Disord., Dept. of Surgery, Univ. of Louisville, Louisville, KY), and William M. Hartmann (Dept. of Phys. and Astronomy, Michigan State Univ., East Lansing, MI)

Binaural squelch is a perceptual phenomenon whereby the subjective strength of reverberant sound is attenuated under binaural listening conditions relative to monaural or diotic listening conditions. Although the effect is well known, only a few studies have explicitly quantified the effect and all used uni-dimensional objective measures such as word recognition or echo detection. Here, multidimensional scaling techniques were used to more directly quantify the perceptual attributes of binaural squelch. In a series of experiments, listeners were asked to rate the perceptual similarity between pairs of sound sources that varied in distance (1–12 m), level (± 3 dB), and presentation mode (binaural versus diotic). Sounds were presented using virtual auditory space techniques to simulate sound-field listening to a frontal source in a reverberant room (broadband T60 = 2 s) and in anechoic space. The source signal was a brief sample of speech from a female talker. Multidimensional scaling (INDSCAL) results revealed that effects of binaural squelch are most evident on a perceptual dimension strongly related to the ratio of direct-to-reverberant sound energy, although individual differences were observed. Additional effects, unrelated to sound level, were also evident on a second perceptual dimension.

1pPP36. Measures of ear lateralization in a dichotic sample discrimination task. Alison Tan and Bruce Berg (Cognit. Sci., UC Irvine, 2201 Social & Behavioral Sci. Gateway Bldg., Irvine, CA 92617, aytan@uci.edu)

This study investigates the ability to lateralize using a dichotic sample discrimination task. On each trial, seven 60-ms tones are drawn from a normal distribution with means of 1000 or 1100 Hz. Even numbered tones are the most informative ($d' = 2$) and presented to one ear and the less informative, odd numbered tones ($d' = 0.5$) are presented to the other ear. Participants indicate from which distribution the tones are sampled. Task difficulty is manipulated by presenting odd-and even-numbered tones at different intensities. In easier conditions, informative and less informative tones are presented at 70 dB, 50 dB, respectively. In difficult conditions, informative and less informative tones are presented at 50 dB, 70 dB, respectively. Decision weights, efficiency measures, and performance level (d') all show an unexpectedly wide range in a listener's ability to lateralize and attend to the most informative tones. Estimates of d' range from 2.4 to 0.7. Some listeners display high efficiency estimates in all conditions while others show a marked ear preference, achieving high efficiency only when the informative tones are presented to a particular ear. Another group of listeners show a distinct inability to lateralize in any condition. This latter group is most affected by the intensity manipulation.

Session 1pSAa

Structural Acoustics and Vibration, Noise, and Signal Processing in Acoustics: Blind Source Localization and Separation for Structural Acoustic Sources

Piervincenzo Rizzo, Chair
University of Pittsburgh, 942 Benedum Hall, 3700 O'Hara Street, Pittsburgh, PA 15261

Chair's Introduction—1:00

Invited Papers

1:05

1pSAa1. Source localization and signal extraction using spherical microphone arrays. Mingsian R. Bai and Yueh Hua Yao (Power Mech. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu, Taiwan 30013, Taiwan, msbai63@gmail.com)

Source localization and extraction using spherical microphone arrays is presented in this paper. Both freefield and solid spherical arrays are examined. To minimize the basis mismatch problem in signal extraction, source localization is executed first using two beamforming methods, delay and sum (DAS) and minimum variance distortionless response (MVDR). The beamformers are formulated in both the spatial domain and the modal domain. After source localization, Tikhonov regularization (TIKR) and compressive sensing (CS) are used to extract the source amplitude signals. Simulations and experiments are conducted for spherical arrays of radius 5 cm with 32 microphones mounted on the vertices and faces of an icosahedron. The results demonstrate better localization performance of the spatial beamformer that the modal beamformer. In addition, the source speech signals are extracted by using the proposed arrays with superior quality.

1:30

1pSAa2. Integrating Markov random fields and model-based expectation maximization source separation and localization. Michael I. Mandel (Comput. Sci. and Eng., The Ohio State Univ., Columbus, OH) and Nicoleta Roman (The Comput. Sci. and Eng., The Ohio State Univ., Lima, 4240 Campus Dr., Lima, OH 45804, roman.45@osu.edu)

Separation of multiple sources from binaural reverberant audio mixtures has been shown to be a challenging task. Binaural algorithms typically perform separation by employing localization based clustering or classification in individual time-frequency units. By assuming that a single source is active in a particular time-frequency unit, these algorithms produce spectral masks that allow the separation of an arbitrary number of sources. The model-based EM Source Separation and Localization (MESSL) algorithm increases robustness to reverberation by explicitly modeling its effects on interaural phase and level differences and using a consistent estimate of interaural time difference across all frequencies. The current study has extended the MESSL algorithm with a Markov Random Field based mask smoothing procedure that enforces consistency in the assignment of neighboring time-frequency units to sources. The proposed MESSL-MRF algorithm is tested in highly reverberated conditions and shows significant improvements over the original MESSL algorithm as measured by both signal-to-distortion ratios as well as a speech intelligibility predictor.

1:55

1pSAa3. Impulsive sound localization using time-domain beamformer. Dae-Hoon Seo, Jung-Woo Choi, and Yang-Hann Kim (School of Mech., Aerosp. and System Eng., Korea Adv. Inst. of Sci. and Technology(KAIST), 373-1 Guseong-dong, Yuseong-gu, Daejeon 305-701, South Korea, ihuny@kaist.ac.kr)

This paper presents a beamforming technique for locating impulsive sound source in low signal to noise ratio (SNR). Impulsive sound generates a high peak-pressure for a short duration in time-domain. However, the measured signal of the sensor are always embedded in the interference noise sources and measurement noise, therefore, it is difficult to estimate the direction of arrival of impulse sound in low SNR environment. In contrast to a frequency-domain beamformer, it has been reported that a time-domain beamformer can be better suited for transient signals. We proposes peak and crest factor as alternative directional estimators to enhance the performance of a time-domain beamformer in view of the fact that impulsive sound has high peak sound pressure as well as short duration time compared to the ambient noise sources. The performance of three directional estimators, the peak, RMS, and crest factor of output values, are investigated and compared with the incoherent measurement noise embedded in multiple microphone signals. The proposed formula is verified via experiments in an anechoic chamber using a uniform linear array, and the results show that the crest factor estimation of beamformer output determines the direction in a low SNR condition in which interference sources are dominants.

Contributed Paper

2:20

1pSAa4. Automating source localization and separation using sonic detection and ranging. Yazhong Lu (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, tiduslyz_01@hotmail.com)

The sonic detection and ranging (SODAR) technology developed previously (Wu and Zhu, JASA, 2013) is automated for performing blind sources localization and separation. In particular, source localization results are displayed in terms of space-frequency or space-time correlations. In other words, one can either view distributions of sound sources in three-dimensional (3-D) space versus any user-defined frequency bands, or distributions of sound sources in 3-D space versus time history. The sound pressure level (SPL) values associated with the identified sources are also calculated and displayed in these space-frequency and space-time correlation graphs. To acquire a better understanding of the distribution of sound sources together with their SPL values in 3-D space, a 3-D viewer using Google SketchUp software is employed to view these results in both space-frequency and space-time correlations. With this 3-D viewer, one is able to rotate and look at any source from any perspective, and zoom in and out to examine the details of relative positions of individual sources. The information on source locations together with windowing and filtering technologies enable us to separate the individual source signals. Examples of using this blind sources localization and separations in a non-ideal environment that involves random background noise and unspecified interfering signals are demonstrated.

MONDAY AFTERNOON, 18 MAY 2015 — KINGS 3, 3:00 P.M. TO 5:00 P.M.

Session 1pSAb

Structural Acoustics and Vibration: General Topics In Structural Acoustics and Vibration I

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

Contributed Papers

3:00

1pSAb1. Prediction of broadband high-frequency acoustic reflection and transmission from subcritical membranes and elastic plates. Mauricio Villa and Donald B. Bliss (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., 148B Eng. Bldg., Durham, NC, donald.bliss@duke.edu)

Earlier work has shown that distributed broadband excitation of subcritical membranes and plates can yield surprisingly simple closed-form band-averaged directivity patterns, and that the directivity is closely related to energy flow in the structure. A first principles energy-intensity boundary element method (EIBEM) for sound fields in enclosures has also been developed, and shown to be very accurate for impedance reflection boundary conditions. The present work involves the derivation of more realistic reflection and transmission of models for enclosure boundaries with elastic boundaries. For subcritical panels, reflection, radiation, and transmission properties can be characterized by a limited set of parameters, and fairly simple analytical results are derived for the broadband case. Surface reflection and edge re-radiation are both considered. These results facilitate an energy-intensity reformulation of the structural equations, as well as the acoustic field, allowing for the possibility of formalizing the coupling between energy flows in the acoustic and structural systems. The goal is to develop efficient first principles computational models for more accurate modeling of real enclosures.

3:15

1pSAb2. Detecting building leakages using nearfield acoustic holography technique: A numerical simulation. Hirenkumar J. Patel, Kanthasamy Chelliah, Ganesh Raman (Mech. Mater. and Aerosp., Illinois Inst. of Technol., 10 W 32nd St., E1- Ste. 243, Chicago, IL 60616, hpatel58@iit.edu), Ralph T. Muehleisen, and Eric Tatara (Argonne national Lab., Lemont, IL)

A crack on a building wall and sound generated inside the building using an artificial sound source were simulated using Matlab. Various noise types affecting the pure sound signal, such as background noise in the room, were simulated. The forward problem of sound propagation from the crack surface to a virtual microphone array was simulated using the free-space green's function. Nearfield acoustic holography (NAH) was applied to perform the inverse problem to calculate the sound pressure at the crack surface and determine the crack location on the building wall. Effects of various errors and noise on the sound pressure reconstruction using NAH were studied. The reconstructed sound pressure levels on the wall surface were compared with the originally simulated sound pressure data at the crack surface. Various crack sizes and shapes were simulated to determine the correlation between the size of the crack and the reconstructed pressure field.

3:30

1pSAb3. Modal response of a simply supported plate driven by a single, asymmetric piezoelectric actuator for use as a flat-panel loudspeaker. Michael C. Heilemann and Mark F. Bocko (Elec. and Comput. Eng., Univ. of Rochester, 500 Joseph C Wilson Blvd., Rochester, NY 14627, mheilema@ur.rochester.edu)

An approximate analytical solution for the mechanical response of a thin plate to a single, asymmetrically bonded piezoelectric driver has been found. Previous work has focused only on the response of plates driven by symmetrically aligned actuators or for beams with drivers bonded to one side. The relative magnitudes of extensional and flexural waves produced by the single-sided driver configuration were determined as a function of the physical parameters of the plate and driver. The modal response of flexural waves for a simply supported thin plate also was determined. Interactions at the plate boundaries with the support structure may convert extensional waves into sound-producing flexural waves, which can lead to audio distortion. A non-moment inducing support scheme with high damping is proposed to minimize this effect.

3:45

1pSAb4. Characterization of the dominant structural vibration of hearing aid receivers. Brenno Varanda (Mech. Eng., Binghamton Univ., 1410 Alexander Way, Clearwater, FL 33756, bvarand1@gmail.com), Ron N. Miles (Mech. Eng., Binghamton Univ., Vestal, NY), and Daniel Warren (Specialty Components - Acoust., Knowles Corp., Itasca, IL)

The overall aim of this research is to analyze and characterize the mechanical vibration of hearing aid receivers, a key electro-acoustic component of hearing aids. The receiver is a high efficiency miniature sound source which utilizes a balanced armature electromagnetic motor. A standard side effect for most balance armature receivers is structural vibration. This receiver-borne structural vibration can travel through the hearing aid package to the microphones, resulting in undesirable oscillations, just like acoustic feedback. To better understand and control this important source of feedback in hearing aids, a simple dynamic model has been developed to describe the system. The model consists of two rigid bodies connected by a torsional spring and damper. A method was developed to estimate the parameters for the dynamic model using experimental data. The data were collected using translational velocity measurements using a scanning laser vibrometer of a Knowles ED-series receiver on a complaint foundation. The analytical dynamic model was validated with finite element analysis using COMSOL and the multibody dynamics module.

4:00

1pSAb5. Active and passive monitoring of valve bodies utilizing spray-on transducer technology. Kenneth R. Ledford (Acoust., Penn State, 309B EES Bldg. Penn State, University Park, PA 16802, Krl175@psu.edu), Kyle Sinding, and Bernhard Tittmann (Eng. Sci. and Mech., Penn State, University Park, PA)

Structural health monitoring (SHM) and non-destructive evaluation (NDE) can be performed both actively and passively. Active monitoring is useful for thickness measurement and crack interrogation. Passive monitoring can indicate integrity of the valve and if it is open or closed. Traditional SHM methods for valve bodies require a bonding medium that can deteriorate. A sol-gel spray-on technology eliminates the need for a coupling medium since the transducer is chemically bonded to the valve body. These spray-on transducers can be tailored to specific applications in order to maximize response or operating temperature. This technology allows for efficient on-line monitoring of valve bodies for thickness and valve integrity. The current objective is to develop a spray-on transducer and corresponding method for both active and passive SHM of valve bodies. The objective relating specifically to passive monitoring relates to indicating the condition of the valve and determining what position it is in. The active monitoring goals are to measure the thickness of the valve in critical regions, determine the presence of corrosion, and observe the roughness of the interior surface. This paper provides preliminary results on the use of spray-on transducers for on-line SHM of valve bodies for both active and passive applications.

4:15

1pSAb6. Reconstructing acoustic field based on the normal surface velocity input data. wu zhu and Sean F. Wu (Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., 2100 Eng. Bldg., Detroit, MI 48202, es1699@wayne.edu)

Traditional NAH relies on the acoustic pressure collected on a hologram surface in the near field of a vibrating structure. To ensure accuracy in reconstruction, it is necessary to have a conformal array of microphones, which is not an easy task in practice. On the other hand, it is easy to use a scanning laser vibrometer to measure the normal surface velocity. This paper presents a modified Helmholtz Equation Least Squares (HELS) formulation to reconstruct the acoustic field based on the normal surface velocity input data. This approach is advantageous in that: (1) it enables one to collect input data in the far field; (2) measurement setup is very simple; and (3) the normal surface velocity contains all near-field information for reconstruction of an acoustic field. To ensure the accuracy in reconstruction, the normal surface velocity is supplemented by a few measurements of the acoustic pressure in the far field. With this combined input data, the acoustic field in three-dimensional space can be accurately reconstructed. Numerical simulations for reconstructing an acoustic field generated by arbitrarily shaped objects are demonstrated. Experimental validations on using this modified HELS formulation to reconstruct the acoustic field from a loudspeaker are also presented.

4:30

1pSAb7. Influence of ground impedance on the sound radiation of a railway sleeper. Xianying Zhang, Giacomo Squicciarini, and David J. Thompson (ISVR,Univ. of Southampton, Southampton SO171BJ, United Kingdom, xz24g12@soton.ac.uk)

A railway track consists of rails attached to sleepers (cross ties) which are laid in ballast. The sleeper provides support for the rail and transfer loads to the ballast and subgrade. Due to the wheel/rail interaction the rail is induced to vibrate and this vibration is transmitted to the sleepers; both the rail and the sleepers radiate sound. Existing models used to predict the sound radiation from the sleeper consider this to be completely embedded in a rigid ground; in reality, however, the sleeper is surrounded by, or embedded to some extent, in the ballast. It is therefore necessary to take these conditions into account in order to obtain a more realistic model. This paper investigates the influence of the ground in close proximity to the sleeper on its sound radiation. A 1/5 scale concrete sleeper is analyzed by using the boundary element method in 3-D. Ground absorption is introduced in terms of its acoustic impedance using the Delany-Bazley model and its effects on the sleeper radiation are predicted. Finally, the numerical results are validated by experimental results using a 1/5 scale model.

4:45

1pSAb8. Measurement of the moisture content of a waste paper bale based on the impact resonance test. Minho Song, Donghyun Kim, Won-Suk Ohm (Mech. Eng., Yonsei Univ., Seoul, Korea, Republic of, Yonsei University, Yonsei-Ro 50, Seodaemoon-Gu, Eng. Bldg. A391, Seoul 120-749, South Korea, songmh@yonsei.ac.kr), and Baek-Yong Um (Balance Industry Co.,Ltd., Seoul, South Korea)

The quality of a waste paper bale depends heavily on its moisture content. The higher the moisture content, the lower the quality as a recycled resource. Therefore, accurate measurement of the moisture content of a waste paper bale is a pressing concern in the paper recycling industry. In this paper, a theoretical model for a waste paper bale, an exotic and complex medium, is developed, in which the acoustic properties of the bale are assumed to be functions of the moisture content. The model is validated through a series of impact resonance tests, which show a strong correlation between the moisture content, resonance frequencies, and the associated quality factors.

Session 1pSC

Speech Communication and Psychological and Physiological Acoustics: Listening Effort II

Alexander L. Francis, Cochair
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Christian Fullgrabe, Cochair
Institute of Hearing Research, Medical Research Council, Science Road, Nottingham NG7 2RD, United Kingdom

Chair's Introduction—1:30

Invited Papers

1:35

1pSC1. Modeling the balance between bottom-up and top-down processing in speech intelligibility tests. Verena N. Uslar, Thomas Brand, and Birger Kollmeier (Medical Phys. and Acoust., Univ. of Oldenburg, Marie-Curie-Strasse 2, Oldenburg 26129, Germany, verena.uslar@uni-oldenburg.de)

Understanding speech depends on sensory, input-driven bottom-up processing as well as on top-down processing-based, e.g., on experience—which is typically associated with more effortful listening. A qualitative and quantitative analysis of the respective contribution of both processing types is important to further diagnostics of hearing impairment, rehabilitation with hearing devices, and the predictive quality of speech intelligibility models. This talk gives a short overview of studies, which measured speech intelligibility as a function of linguistic complexity to quantify the respective contribution of both processing types. Comparison of speech reception thresholds for linguistically simple and more complex sentences in varying listening situations for younger and older adults with normal hearing and with hearing-impairment indicated that experience-driven top-down processing becomes more important with increasing listening effort. All listener groups seem to use the same mechanisms to compensate for increasing sensory or cognitive load, however the threshold for employing experience-driven top-down processing seems to lower with increasing age and hearing impairment. These results led to the development of the so called "speech processing gauge" presented here which explains the shift of the balance between bottom-up and top-down processing with regards to changes in different variables, i.e., age, hearing impairment, sensory load, and cognitive effort.

1:55

1pSC2. Effects of age, hearing loss, and linguistic complexity on listening effort as measured by working memory span. Margaret K. Pichora-Fuller (Psych., Univ. of Toronto, 3359 Mississauga Rd., Mississauga, Ontario L5L 1C6, Canada, k.pichora.fuller@utoronto.ca) and Sherri L. Smith (Audiologic Rehabilitation Lab., Veterans Affairs Medical Ctr., Mountain Home, TN)

Listeners, especially older or hard-of-hearing individuals, report that understanding conversation in noisy situations is effortful. Individual differences may be due to auditory and/or cognitive processing abilities. It is assumed that auditory processing depends on the individual's degree and type of hearing loss, while cognitive processing depends on the individual's working memory (WM) capacity. Many researchers have measured reading WM span to minimize the effects of hearing loss; however, variations due to hearing loss may be important in assessing processing demands (listening effort) during speech-in-noise understanding. The effects of the acoustical properties of the signal and masker on listening effort have been studied, but less is known about the effects of the linguistic complexity of the materials. We predicted that demands on WM and the correlation between WM span measures and speech-in-noise performance would increase with increasing linguistic complexity, with speech-in-noise performance correlating more strongly with auditory than with visual measures of WM span. To test these hypotheses, we administered speech tests varying in linguistic complexity and measures of both reading and listening WM span. Participants were a group of younger listeners with normal hearing and a group of older listeners with bilateral sensorineural hearing loss.

2:15

1pSC3. Task demands and cognitive abilities impact listening effort for older adult hearing aid users. Stefanie E. Kuchinsky (Ctr. for Adv. Study of Lang., Univ. of Maryland, Maryland Neuroimaging Ctr., 8077 Greenmead Dr., Bldg. #795, College Park, MD 20740, skuchins@umd.edu), Jayne B. Ahlstrom, Emily Franko-Tobin, and Judy R. Dubno (Dept. of Otolaryngol.—Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Understanding speech in background noise requires considerable effort for older adults with hearing loss, even when wearing well-fit hearing aids, but little is known about how effort varies with changes in speech recognition task demands in this population. Following research showing that effort can be indexed by changes in the autonomic pupil dilation response, the current study recorded

pupillometry as older adults wearing hearing aids completed the Words-In-Noise test (e.g., Wilson *et al*., 2007). Using a single loudspeaker at 0 degrees azimuth, participants listened to and repeated monosyllabic words in multi-talker babble at seven signal-to-noise ratios (0 to +24 dB SNR). A nonlinear relationship between word recognition score and pupil dilation was observed, with maximal effort at moderate levels of speech recognition difficulty. Individual differences in cognitive abilities (e.g., working memory, vocabulary knowledge) modulated this pattern of results. These findings highlight an important dissociation between listening effort and speech recognition task difficulty. Implications for evaluating changes in effort with speech-perception training are discussed. [Work supported, in part, by NIH/NIDCD.]

2:35

1pSC4. Speech recognition and listening effort across various conditions in adults with aphasia. Carolyn Richie (Commun. Sci. & Disord., Butler Univ., 4600 Sunset Ave., Indianapolis, IN 46205, crichie@butler.edu)

Some of the common complaints among adults with aphasia are that it is difficult to understand speech in noise and that listening can be very effortful. Nonetheless, there has been limited research to date on speech recognition in noise or listening effort for adults with aphasia. Stanfield & Richie (2014) investigated speech recognition under various listening conditions for adults with non-fluent aphasia. Participants with aphasia performed better in quiet as compared to noise, as expected, and better in noise that consisted of multi-talker babble as compared to one competing talker. They also showed modest benefit from the addition of visual cues to speech over auditory-only speech recognition. However, subjective reports of listening effort were mixed and did not line up with performance on the various tests of speech recognition. In the present study, the relationship between speech recognition under various listening conditions and an objective measure of listening effort, response time, was examined. Participants' word and sentence recognition under auditory-only and auditory-visual conditions, in quiet and in two types of noise, will be reported. The potential clinical significance to better understanding listening effort in adults with aphasia will be discussed as well.

2:55–3:10 Break

3:10

1pSC5. Using dual-task paradigms to assess listening effort in children and adults. Erin Picou and Todd A. Ricketts (Dept. of Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. S, Rm. 8310, Nashville, TN 37232, erin.picou@vanderbilt.edu)

While evidence suggests that hearing aids can reduce listening effort, mixed results are often seen when examining the effect of specific hearing aid processing. Given methodological differences across studies, we speculated that not all paradigms are equally sensitive to changes in listening effort. Moreover, many traditional paradigms may not be appropriate for children. The purpose of this project was to evaluate dual-task paradigms for measuring listening effort in adults and school-aged children. Three tasks were developed based on existing dual-task paradigms and modified to be appropriate for children. Sixteen adults (aged 22–32) and twenty-two children (aged 9–17) with normal hearing participated. All participants were tested in quiet and in noise. For all three paradigms, the primary task was monosyllable word recognition and the secondary task was a physical response time measure. The secondary tasks varied in complexity or depth of processing. Results revealed that the paradigm requiring deeper processing was most sensitive to the effects of noise for adults, but for children, the paradigm with the highest complexity was the most sensitive. Potential explanations for these group differences and the application of the most sensitive measures for future investigations will be discussed. [This work was funded by Phonak.]

3:30

1pSC6. Subjective listening effort. Michael Schulte, Melanie Krüger, Markus Meis, and Kirsten C. Wagener (Hörzentrum Oldenburg, Marie-Curie-Str. 2, Oldenburg 26129, Germany, m.schulte@hoerzentrum-oldenburg.de)

When investigating listening effort the individual subjective feedback is important to learn about and take into account the situations in which hearing impaired suffer in terms of listening effort and the relationship of listening effort to speech intelligibility. Here, we present results from a daily life questionnaire study and results from a new tool to evaluate listening effort subjectively under controlled conditions in the lab. A research questionnaire was used to determine the individual subjective listening effort in daily life in a systematic way. The questionnaire with 29 well described situations was used in a multi-center study in Denmark, Germany, and USA with 112 subjects to identify the most relevant effortful situations (e.g., "watching news in TV" or "talking at the kitchen table"). As a lab procedure we developed an adaptive categorical listening effort scaling procedure. It is based on a non-adaptive version that turned out to be a sensitive lab method in which subjects rate listening effort at predefined signal to noise ratios. The new method automatically finds the SNRs corresponding to subjective ratings from "extreme effort" to "no effort." First results show the relationship between listening effort and intelligibility for different SNRs and background noises.

Contributed Papers

3:50

1pSC7. Verbal response time in children systematically varies with the spectral resolution of speech. Kristi M. Ward, Jing Shen, Pamela E. Souza, and Tina M. Grieco-Calub (The Roxelyn & Richard Pepper Dept. of Commun. Sci. & Disord., Northwestern Univ., 2240 N. Campus Dr., Rm. 2-381, Evanston, IL 60208, kmward@u.northwestern.edu)

Children require higher spectral resolution to perform comparably to adults on speech recognition tasks. A potential source of this age-related difference is children's immature cognitive processing. Verbal response time (VRT), the speed at which listeners verbally repeat speech stimuli, has been used as a correlate of cognitive load (or listening effort, LE) in children: listeners exhibit faster VRT in conditions with greater signal-to-noise ratios and slower VRT in conditions with decreased audibility. If VRT is representative of changing cognitive demands during speech recognition, and therefore changing LE, in children, we predict that gradually decreasing the spectral resolution of the speech signal will result in a concomitant slowing of VRT. To test this prediction, we measured VRT in typically developing children (8–12 years old) while they performed a sentence recognition task that was either unprocessed (full spectral resolution) or noiseband vocoded

with 4, 6, or 8 spectral channels. Preliminary data suggest that children's VRT varies systematically with spectral resolution: VRT slows with decreasing spectral resolution of the speech signal. These results are consistent with the idea that VRT is a sensitive measure of LE in children. Discussion will consider these data in the context of other methods of measuring LE.

4:05

1pSC8. Honking is just noise (or just about): The effect of energetic masking on recognition memory for spoken words. Dorina Strori (Dept. of Psych., The Univ. of York, York YO10 5DD, United Kingdom, dorina.strori@york.ac.uk), Johannes Zaar (Tech. Univ. of Denmark, Copenhagen, Denmark), Odette Scharenborg (Radboud Univ. Nijmegen, Nijmegen, Netherlands), Martin Cooke (Univ. of the Basque Country, Vitoria, Spain), and Sven Mattys (The Univ. of York, York, United Kingdom)

Previous research indicates that listeners encode both linguistic and indexical specifications of the speech signal in memory. Recent evidence suggests that non-linguistic sounds co-occurring with spoken words are also incorporated in our lexical memory. We argue that this "sound-specificity effect" might not be due so much to a word-sound association as to the different acoustic glimpses of the words that the associated sounds create. In several recognition-memory experiments, we paired spoken words with one of two car honk sounds and varied the level of energetic masking from exposure to test. We did not observe a drop in recognition accuracy for previously heard words when the paired sound changed as long as energetic masking was controlled. However, when we manipulated the temporal overlap between words and honking to create an energetic masking contrast, accuracy dropped. The finding suggests that listeners encode irrelevant non-speech information in memory, but only in certain contexts. Calling for an expansion of the mental lexicon to include non-speech auditory information might be premature. Current work is investigating the effect in non-native listeners of English, and whether maskers that are more integral to the words and hence more difficult to segregate lead to a more robust effect.

4:20–5:05 Panel Discussion

MONDAY AFTERNOON, 18 MAY 2015 KINGS 5, 1:00 P.M. TO 4:30 P.M.

Session 1pSP

Signal Processing in Acoustics, Architectural Acoustics, and Noise: Telecom and Audio Signal Processing

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

Invited Papers

1:00

1pSP1. Microphone cross-array beamformer processing to reduce noise and reverberation. Gary W. Elko (mh Acoust., 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com), Jens Meyer (mh Acoust., Fairfax, Vermont), and Steven Backer (mh Acoust., Oakland, California)

Hands-free audio communication systems that are designed to allow audio and speech communication between remote parties are known to be sensitive to room reverberation and noise when the source is distant from the microphone. A common solution to the problem of room reverberation is to use an array of microphones to spatially filter the acoustic field. The maximum array gain is only attainable with specific microphone geometries and the gain of realizable microphone arrays is typically significantly lower than the maximum. The algorithm described in this talk attempts to address the rather slow growth in linear processing directional gain as a function of the number of microphones. One possible approach to attain higher directive gain is to use a processing scheme based on nonlinear multiplicative processing. An obvious candidate is the coherence function since it is a bounded and normalized multiplicative measure. A technique is presented for reverberation reduction processing using at least two beamforming microphones and a function of the estimated short-time coherence between the beamformer outputs. Each beamformer should have a different directional response or spatial position, or both, but with overlapping responses in the direction of the desired source.

1:20

1pSP2. Study and design of robust differential beamformers with linear microphone arrays. Jacob Benesty (INRS-EMT, Univ. of PQ, Montreal, Quebec, Canada), Jingdong Chen, Chao Pan, and Hao Zhang (Ctr. of Intelligent Acoust. and Immersive Communications, Northwestern PolyTech. Univ., 127 Youyi West Rd., Xi'an, Shaanxi 710072, China, jingdongchen@ieee.org)

Differential beamformers can generate frequency-invariant spatial responses and therefore have the great potential to solve many broadband acoustic signal processing problems such as noise reduction, signal separation, dereverberation, etc. The design of such beamformers, however, is not a trivial task. This paper is devoted to the study and design of differential beamformers with linear array geometry. The objective is to design robust differential beamformers that can form frequency-invariant beampatterns. The major contribution consists of the following aspects. (1) It discusses a general approach to the design of linear DMAs that can use any number of microphones to design a given order DMA as long as the number of microphones is at least one more than the order of the DMA. (2) It presents a method that can maximize the white noise gain with a given number of microphones and order of the DMA; so the resulting beamformer is more robust to sensor noise than the beamformer designed with the traditional DMA method. (3) It discusses how to use nonuniform geometries to further improve robustness of differential beamformers. (4) It investigates the possibility to improve the robustness of differential beamformers with the use of the diagonal loading technique.

1:40

1pSP3. Study of the maximum signal-to-noise-ratio filters for single- and multi-channel noise reduction. Jingdong Chen (INRS-EMT, Univ. of PQ, 127 Youyi West Rd., Xi'an, Shaanxi 710072, China, jingdongchen@ieee.org), Gongping Huang (Ctr. of Intelligent Acoust. and Immersive Communications, Northwestern PolyTech. Univ., Xi'an, Shaanxi, China), and Jacob Benesty (INRS-EMT, Univ. of PQ, Montreal, Quebec, Canada)

Noise reduction is a problem of recovering a speech signal of interest from its noisy observations. Since the objective of the problem is to reduce noise, thereby improving the signal-to-noise ratio (SNR), it is natural to consider the use of maximum SNR filters. However, the maximum SNR filters, if not designed properly, may introduce significant speech distortion, leading to speech quality degradation instead of improvement. This paper investigates the design of maximum SNR filters for noise reduction with minimum speech distortion. It covers the following design cases: (1) single-channel noise reduction in the time domain; (2) multichannel noise reduction in the time domain; (3) single-channel noise reduction in the short-time-Fourier-transform (STFT) domain with or without interframe information; (4) multichannel noise reduction in the STFT domain with or without interframe information. A large number of experiments are performed to illustrate the properties of these maximum SNR filters.

2:00

1pSP4. Coded excitation in space and time for the identification of geometric parameters using audio transducer arrays. Jung-Woo Choi (Mech. Eng., KAIST, 291 Daehak-ro, Yuseong-gu, Daejeon 305-701, South Korea, jwoo@kaist.ac.kr)

Impulse responses of a room can provide a lot of geometric information of acoustic systems. Loudspeaker and microphone array positions, locations of walls and scattering bodies are representative examples. To extract geometric information from the room impulse responses, the measurement time should be short enough such that the degradation of time delay estimation performance due to non-stationary environmental noises can be minimized. In this work, we attempt to enhance the SNR of the short-term measurement, by using a coded excitation technique in space and time. Both the spatially and temporally coded excitation signals are applied to a set of loudspeakers combined with microphones, and it is shown that the measurement time can be shortened with minimal loss of SNR and time delay estimation performance.

2:20

1pSP5. Adaptive beamforming for acoustic echo cancellation using measured array models and subband filtering. Mingsian R. Bai and Li-Wen Chi (Power Mech. Eng., National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu, Taiwan 30013, Taiwan, msbai63@gmail.com)

Acoustic echo that can substantially undermine speech quality is one of the key issues must be addressed in practical telecommunication systems. In this paper, an evolutionary exposition is given in regard to the enhancing strategies for acoustic echo cancelers (AEC). On the basis basic adaptive filtering, a highly directional microphone array is designed as a fixed beamformer (FBF) to focus on the near-end speaker and suppress the echo from the far-end as well as noise and interference from the background. Design of the beamformer is based on array models interpolated from measured frequency responses. Subband (SB) filtering with polyphase decomposition is exploited to accelerate the cancellation process with multiple choice of step size in the subbands. To further enhance the canceler, adaptive generalized sidelobe canceler (GSC) can be utilized, which is comprised of a FBF for the near-end speech and an adaptive blocking module for the echo and noise. Objective tests in terms of echo return loss enhancement (ERLE) and perceptual evaluation of speech quality (PESQ) are conducted to compare four algorithms with various combinations of enhancement. The results show that the GSC-SB-AEC approach has attained the highest ERLE and best speech quality, even in a double-talk scenario.

2:40–3:00 Break

Contributed Papers

3:00

1pSP6. Experimental verification of a Fisher Information Model for azimuth estimation in broadband active sonar. Colin J. Ryan (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 679 Main St., Harwich, MA 02645, cryan2@umassd.edu), John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, North Dartmouth, MA), and Laura N. Kloepper (Biology, Brown Univ., Providence, RI)

Bats easily identify targets of interest and reject clutter in the wild. A bat's transmit and receive beampatterns are frequency dependent, so it is possible that they exploit spectral cues to distinguish on-axis targets from off-axis clutter. The frequency dependent beampattern can also be interpreted as a different lowpass filter for each angular arrival. Given the known transmitted signal, the angle of arrival can be estimated from the high frequency attenuation in the received signal. The broadband Fisher information (FI) for azimuth in active sonar predicts a maximum FI at a slightly off-axis location, suggesting bats might target their sonar beam askance of a target to maximize the precision in estimating azimuth. This paper tests the FI model in a laboratory with a baffled tweeter as the sonar transmitter and an omnidirectional receiver. The experimentally measured beampatterns are used in Monte Carlo simulations with a maximum-likelihood estimator for azimuth over a range of SNR. The experimental results gained from the laboratory tests are compared with the model to assess the agreement between the predicted azimuth of maximum FI and the observed azimuth minimizing experimental error variance. [Funded by UMass Dartmouth Office of Undergraduate Research, College of Engineering, and ONR.]

3:15

1pSP7. Developing an audio network system for teleconferencing with evaluation of audio quality. Hansol Lim, Hyung Suk Jang, Muhammad Imran, and Jin Yong Jeon (Architectural Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, sollim0128@gmail.com)

A teleconference system was developed to transmit speech signals for audio communication with remote users. The hardware system was composed of 3-D microphone arrays to capture directional sound and a wireless headset to provide freedom of movement for users. An application programming interface (API) was used for interaction between the computer and the audio devices, including the microphone, headphone, and ADDA converter, transferring input/output audio signals in real time through the network. For the evaluation of the system, a latency test was performed with a simultaneous recording system in capturing and reproduction positions. The latency from the network procedure was calculated with different buffer sizes, and the lower-delay and smaller-deviation buffer was selected. In addition, the transmitted audio signals were compared with the input signals in terms of signal patterns and frequency responses. The resulting audio qualities in the time and frequency domain were suitable for a teleconferencing system.

3:30

1pSP8. Tone-to-text applications for sub-Saharan African tone languages. Aaron Carter-Enyi (The Ohio State Univ., 2180 N 4th St., Columbus, OH 43201, cartercohn@gmail.com)

Based on experimental findings, this paper proposes a framework for computer recognition of speech tones in Niger-Congo languages. This language family includes over 1000 languages in Sub-Saharan Africa and approximately one billion speakers. Many of these languages use pitch contrast to differentiate words in a system of two or more pitch levels (e.g., high, mid, and low). The results of two new studies conducted in Nigeria from 2013-14 indicate speech-tone is perceived as tonemic intervals (e.g., high-high, high-mid, and high-low). An experimental study (n = 1448) identified ranges of pitch difference that form perceptual categories for these tonemic intervals. An initial application of these findings is adding tone diacritics to text by interpreting fundamental frequency. Representing tone in text has been a persistent problem for Niger-Congo languages and smartphones are ill-equipped for marking tone diacritics. A signal processing application that recognizes speech tones and marks them on computer text would be more efficient than manually highlighting and marking each syllable. Frequency distribution of tones is dynamic between speakers. Thus, there are potential benefits of using machine learning to create speaker-dependent software. Because the proposed method relies on existing algorithms for fundamental frequency, the problem of estimation errors will also be addressed.

3:45

1pSP9. Subarray based localization using extended coprime arrays. Andrew T. Pyzdek, R. Lee Culver, and Dave Swanson (Appl. Res. Lab., The Penn State Univ., PO Box 30, State College, PA 16804, atp5120@psu.edu)

Partitioning of long linear arrays into a number of smaller subsections, termed subarrays, is a common form of processing used both to compensate for irregularity in array shape and to localize near-field sources. While this manner of processing is applicable to uniform linear arrays, as the subarrays are similarly uniform and possess the same minimum element spacing, subarray methods cannot always be applied to sparse arrays with uneven spacing along their length. One sparse array design of interest is the extended coprime array, an array composed of two uniformly spaced component arrays, each undersampled by integer factors which are selected to be coprime. By exploiting the regularity of spatial lag repetitions in extended coprime arrays, we show that appropriately selected subsections of a coprime array can be used as subarrays to determine distance to a near-field source. The performance of the coprime array processed in this manner will be compared to similar processing performed on alternative sparse array designs and the baseline performance of a uniform linear array of equal aperture.

4:00

1pSP10. Testing spatial co-prime sampling theory. Radienxe Bautista and John R Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, rbautista@umassd.edu)

An array of sensors can be used to estimate the direction of arrival of a narrowband signal in the far field with the use of conventional beamforming. In order to avoid spatial aliasing, the distance between the sensors have to be $d \leq \lambda/2$ apart. This is analogous to the Nyquist theorem for sampling in time with $f_s = 2f_o$. Co-Prime arrays are non-uniform arrays that can predict what a uniform array with an aperture of $L = M \times N$ total sensors can using fewer sensors; they have a total number of sensors $L = M + N—1$ while maintaining an aperture of $M \times N$ sensors in a full uniform line array with $\lambda/2$ spacing between elements. Each subarray is uniform linear, equally spaced by $N\lambda/2$ or $M\lambda/2$ and then combined to create a non-uniform linear array. The result with the different inter-element spacing between the subarrays is grating lobes in different locations with the exception of the true DOA. Conventional beamforming is done with both subarrays and the outputs are then multiplied together to yield a single beam pattern. The theory to this will be tested using a uniform array of 30 sensors and selecting the data from certain sensors to achieve co-prime sampling.

4:15

1pSP11. The study of adaptive time reversal mirror as a crosstalk mechanism for underwater communication. Yi-Wei Lin (Systems and Naval Mechatronics Eng., National Cheng Kung Univ., No.1, University Rd., Tainan 701, Taiwan, ls5028@gmail.com)

The time reversal mirror technique has been widely applied to mitigate the inter-symbol interference in underwater channels. Meanwhile, an adaptive time reversal mirror is introduced to improve the crosstalk quality between receivers in underwater communication. To explore the effectiveness of this method, this study extended the analysis of adaptive time reversal mirror as a crosstalk mechanism and explored the mechanism in an experiment using a towing tank as a testing platform. The advantage of this process is its simplicity in examining the effects of the array configuration of this crosstalk mechanism. Results of parametric experiments are discussed i.e. the effects of number of receivers, spacing between sources and noise energy threshold. Experimental data at 10 and 16 kHz with a 5-kHz bandwidth demonstrate as much as an 8-dB signal to noise ratio improvement for four receivers dual sources over a 30 m communication range in a 3.3 m depth testing platform. The results indicate array configuration affects this mechanism drastically.

Session 1pUW

Underwater Acoustics: Environmental Characterization, Localization, and Vectors

David J. Zartman, Cochair
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Marine Physical Laboratory, Scripps Institution of Oceanography, 9500 Gilman Drive, La Jolla, CA 92093

Contributed Papers

1:00

1pUW1. Array shape calibration using long-term ambient noise records. Stephen Nichols and David L. Bradley (Appl. Res. Lab., The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, smn5198@psu.edu)

When using an acoustic array to determine the bearing of a source, errors in the sensor positions severely reduce the accuracy of any bearing measurements. This issue is particularly problematic for three sensor arrays, designed to work in two dimensions, because of the lack of redundancy built into the array. This paper presents a method for correcting errors in sensor positions. It is mathematically straightforward to determine the bearing of a far-field impulsive signal by determining the time difference of arrival between sensors by cross-correlation. These time delays can also be used to estimate the sound speed in the vicinity of the array. In an isotropic acoustic medium, the local sound speed is expected to be independent of the source bearing. If the sensor positions used to determine the sound speed and bearing are incorrect, the resulting sound speed measurements will be bearing-dependent. Using an analytically derived function, the correct array shape (with only translational and rotational ambiguity) can be backed out from the directional sound speed distribution. This method will be demonstrated using long-term ambient noise records from the Comprehensive Nuclear-Test Ban Treaty Organization's (CTBTO) hydrophone arrays.

1:15

1pUW2. Maximum entropy inference of seabed attenuation parameters using ship radiatedbroadband noise. David P. Knobles (ARL, Univ. of Texas, PO Box 8029, Austin, TX 78713, knobles@arlut.utexas.edu)

The use of ambient noise as a means of inferring physical properties of the ocean is of significant interest in ocean acoustics. This study employs the received acoustic field generated by the passage of the R/V Knorr on the New Jersey continental shelf to test a new approach that estimates the probability distributions of both the aspect dependent source levels and the parameters that represent the geoacoustic structure of the seabed. Since the source levels and the environmental parameters have an intrinsic ambiguity, both classes of parameters must be included in the model hypothesis space as random variables. The statistics of the error function, needed to uniquely specify the likelihood function, are estimated with a maximum entropy approach by creating a data ensemble that includes samples from time periods where the ship-receiver geometry is dominated by stern, bow, and port aspect. This method has its origins from the observation that Bayes' rule is symmetric with the interchange of the data space and the hypothesis space. [Research supported by ONR 32 OA.]

1:30

1pUW3. High frequency source localization in a shallow ocean sound channel using frequency-difference matched field processing. Brian M. Worthmann and David R. Dowling (Univ. of Michigan, 2010 Lay Automotive Lab, Ann Arbor, MI 48109, bworthma@umich.edu)

Matched field processing (MFP) is an established technique for locating remote acoustic sources in known environments. Unfortunately, environment-to-propagation model mismatch prevents successful application of MFP in many circumstances, especially those involving high frequency signals. For beamforming applications, this problem was found to be mitigated through the use of a nonlinear array-signal-processing technique called frequency difference beamforming (*Abadi et. al*. 2012). Building on that work, this nonlinear technique was extended to Bartlett MFP, where ambiguity surfaces were calculated at frequencies two orders of magnitude lower than the propagated signal, where the detrimental effects of environmental mismatch are much reduced. Previous work determined that this technique has the ability to localize high-frequency broadband sources in a shallow ocean environment with a sparse vertical array, using both simulated and experimental propagation data. Using simulations, the performance of this technique with horizontal arrays and adaptive signal processing techniques was investigated. Results for signals with frequencies from 10 kHz to 30 kHz that propagated in a 100-m-deep shallow ocean sound channel with a downward refracting sound speed profile will be shown for source array ranges of one to several kilometers. [Sponsored by the Office of Naval Research.]

1:45

1pUW4. Passive acoustic source localization using sources of opportunity. Christopher Verlinden, Jit Sarkar, William Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0701, cmverlin@ucsd.edu), Karim Sabra (Mech. Eng. Dept., Georgia Inst. of Technol., Atlanta, GA), and William Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

Rather than use replica fields for matched field processing (MFP) derived from acoustic models requiring detailed environmental input, we demonstrate that data derived replicas from ships of opportunity can be used for assembling a library of replicas for MFP. The Automatic Identification System (AIS) is used to provide the library coordinates for the replica library and a correlation based processing procedure is used to overcome the impediment that the replica library is constructed from sources with different spectra and will be used to locate another source with it own unique spectral structure. The method is illustrated with simulation and then verified using acoustic data from a 2009 experiment for which AIS information was retrieved from the Unites States Coast Guard Navigation Center (USCG NAVCEN) Nationwide AIS (NAIS) database.

2:00

1pUW5. Bayesian linearized two-hydrophone localization of a pulsed acoustic source. Emmanuel Skarsoulis (Foundation for Res. and Technol. - Hellas, Heraklion, Greece) and Stan E. Dosso (Univ. of Victoria, University of Victoria, Victoria, British Columbia, Canada, sdosso@uvic.ca)

A three-dimensional localization method for transient acoustic sources is developed, based on time differences between direct and surface-reflected arrivals at two hydrophones. The method accounts for refraction caused by a depth-dependent sound-speed profile using a ray-theoretic approach, and, further, it provides localization error estimates accounting for uncertainties of the arrival times and hydrophone locations, as well as for depth-dependent uncertainties in the sound-speed profile. In the first of two steps, source depth and range to each hydrophone are estimated using an iterative, linearized Gauss-Markov inversion scheme. In the second step, the estimated source ranges are combined with the hydrophone locations to obtain the source location in the horizontal. Localization performance is analyzed in a simulation study, and the linearized localization estimates and uncertainties are validated by comparison with a fully-nonlinear (but numerically intensive) Markov-chain Monte Carlo inversion. [Work supported by Aristeia-II program, EU-ESF and Greece, NSRF 2007-13.]

2:15

1pUW6. Sound speed structure monitoring of the Antarctic Ocean by new style float. Shinpei Gotoh (Intelligent Interaction Technologies, Univ. of Tsukuba, 1-1-1 Tennodai, Tsukuba, Ibaraki 305-8573, Japan, gotohs@jamstec.go.jp), Toshio Tsuchiya, Yoshihisa Hiyoshi (JAMSTEC, Yokosuka, Kanagawa, Japan), and Koichi Mizutani (Intelligent Interaction Technologies, Univ. of Tsukuba, Tsukuba, Japan)

JAMSTEC developed the new profiling float “Deep NINJA” for deep-sea type, was subjected to long-term monitoring of one year in the Antarctic Ocean off the coast of Adelie Coast from 2012. As a result, succeeded in the long-term monitoring of the sound speed profile to depth of the 4000 m in the Antarctic Ocean for the first time in the world, and was able to capture a seasonal change in the surface area in the freezing season and the thawing season. In addition, calculating a sound speed from these data, simulations were performed assuming the low frequency sonar. The result was obtained, the ingredient which propagates while repeating a reflection in the extremely small layer of the sea surface neighborhood, and the ingredient that propagates while being reflected in near the water depth 100m which changes of sound speed gradient. From this, propagation loss was small in winter than summer, and was shown a possibility that a sound wave would propagate to a more distant place. This may affect the long-distance sound wave propagation of the echo locations of the passive sonar and marine mammals.

2:30

1pUW7. A deep-water experiment to measure mid frequency attenuation. Jeffery D. Tippmann, Jit Sarkar (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0238, jtippman@ucsd.edu), Philippe Roux (Institut des Sci. de la Terre, Universite Joseph Fourier, CNRS UMR 5275, Grenoble, France), William S. Hodgkiss, and William A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California San Diego, La Jolla, CA)

A deep-water experiment was performed off the west coast of the United States with a short vertical array cut for 7.5 Hz and a source transmitting tonals as well as chirps. The data were processed to identify eigenray arrivals out to a convergence zone for the ultimate purpose of revisiting mid frequency, deep water volume attenuation in the Pacific Ocean and comparing it to decades old measurements and models. Further, aside from the beamforming done at the receiver array, attempts were made to construct a synthetic aperture source array in order to perform “double beamforming” and thereby enhance the signal to noise ratio. We present our initial results of this experiment.

2:45

1pUW8. Symmetric and asymmetric reversible quasi-holographic processing of dual transducer sonar for feature extraction and analysis. David J. Zartman, Daniel S. Plotnick, and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Phys. and Astronomy Dept., Pullman, WA 99164-2814, zartman.david@gmail.com)

Two independent transducers in a side-by-side configuration may transmit and/or receive in-phase or out-of-phase with each other. In the transmit mode, the transducers are driven individually; in the receive mode the transducers are recorded independently, allowing the phase relationships to be determined in post-processing. This results in both monostatic and weakly bistatic data. When the transducers are scanned together, as for synthetic aperture sonar, the scattering results from glory target features of a small solid sphere in water are seen to be sensitive to different combinations of in-phase and out-of-phase sources/receivers. Certain combinations function as a spatial derivative and may be generated synthetically from scans using a single transducer. This spatial derivative is shown to be highly sensitive to the alignment of a solid cylinder in water. Imaging via reversible quasi-holographic processing [D. J. Zartman, D. S. Plotnick, T. M. Marston, and P. L. Marston, Proc. Meet. Acoust. **19**, 055011 (2013)] was applied to this target and used for target isolation, feature extraction, identifying and separating system effects from target physics, and for self-normalization. Though transmitting and receiving in-phase improves ordinary SNR, it can reduce some feature sensitivity relative to differential processing. [Work supported by ONR.]

3:00–3:15 Break

3:15

1pUW9. Geo-acoustic inversion of marine sediments using vector sensor measurements. David Dall’Osto (Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@u.washington.edu) and Peter H. Dahl (Appl. Phys. Lab. and Dept. of Mech. Eng., Univ. of Washington, Seattle, WA)

In this paper, a narrow-band geo-acoustic inversion scheme based on the acoustic vector field is presented. Measurements of acoustic particle velocity made with a vector sensor and short line array of hydrophones are studied with regards to the interference pattern generated by single 1–4 kHz source. This source was lowered from a research vessel positioned within three water depths (depth 20-m) of the receivers, and projected a series of 100-ms long continuous wave (cw) pulses as it was raised toward the surface. The interference pattern measured at the receivers is assessed by a non-dimensional property of the acoustic particle velocity, the degree of circularity [Dall’Osto and Dahl, J. Acoust. Soc. Am. **134**, 109 (2013)], computed directly at the vector sensor and approximated along the line-array. This degree of circularity is highly sensitive to the phase of interfering multipaths, which depends on both the source-receiver geometry and the geo-acoustic properties of sea-bed. Source positions measured by GPS and depth sensors are used to model the time-dependent particle velocity field as the cw tone establishes a steady-state response. A best-fit model to the field data provides an estimate of the geo-acoustic properties of the sandy sediment at the experimental site.

3:30

1pUW10. Localization for broadband source by a single vector sensor in shallow water. Junjie Shi, Dajun Sun, Yunfei Lv (Sci. and Technol. on Underwater Acoust. Lab. and College of Underwater Acoust. Eng., Harbin Eng. Univ.,Rm. 1005 of Shuisheng Bldg. of Harbin Eng. University, Harbin 150001, China, junjieshi@hrbeu.edu.cn), and Yun Yu (Naval Acad. of Armament, 100161, Beijing, China)

The physical meaning of array invariant is the absolute traveling time of different modal from a source to a receiver under the average sound speed. It can be gotten by the derivative of the arriving time against the corresponding cosecant function of elevation angle for each modal, which can be used for the passive source localization. A vector sensor can be seen as a small volumetric array. It has its capability of elevation angle determination to describe the multi-modal propagation and the single-modal dispersion in shallow water, which may imply a chance to get the array invariant for the

source localization. Here, we take advantage of the dispersion based short time Fourier transform and warping transform techniques for the time-frequency analysis of vector sensor signals. This technique can improve acoustic normal mode identification and thereby extract the modal data more accurately for array invariant determination. The idea had been effectively validated during the experiment that took place in October 2014 in South China Sea of nearly 100 m depth. [Work supported by the National 863 Project (No. 2011AA090502) and National Natural Science Funds Program (No. 11404406).]

3:45

1pUW11. Study on the interference structure of vector acoustic field in shallow water. Dajun Sun, Junjie Shi, Jidan Mei (Sci. and Technol. on Underwater Acoust. Lab. and College of Underwater Acoust. Eng., Harbin Eng. Univ.,Shuisheng Bldg. of Harbin Eng. University, Harbin, China, junjieshi@hrbeu.edu.cn), and Haizhu Xu (Naval Acad. of Armament, 100161, Beijing, China)

The application of vector sensors is closely associated with the characteristics of vector acoustic field in shallow water, whose study is the foundation for target detection, positioning, and classification. Beginning from Normal Mode theory, theoretical studies are presented about the amplitude of different components of vector acoustic field involving the pressure, horizontal and vertical particle velocity, as well as the phase relation between the pressure and the vertical particle velocity. The main factors considered here are the source depth and the range relative to the vector sensor. These interference structures were observed during the experiment that took place in July of 2013 in South China Sea of nearly 90m depth. The results indicated that the interference structures of vector acoustic field are not only influenced by the sound speed profile but also by seabed parameters including stratification depth and its corresponding acoustic properties. Moreover, the phase relation between the pressure and vertical particle velocity is remarkably impact by these parameters. [Work supported by the National 863 Project (No. 2011AA090502) and Defense Industrial Technology Development Program (B2420132004).]

4:00

1pUW12. Second-order statistics of the instantaneous mutual information in time-varying underwater acoustic particle velocity channels. Chen Chen, Shuangquan Wang (Samsung Mobile Solution Lab, San Diego, CA), and Ali Abdi (New Jersey Inst. of Technol., 323 King Blvd., Newark, NJ 07102, ali.abdi@njit.edu)

Instantaneous mutual information (IMI) is the amount of information that a time-varying channel can convey for the given time instant. In this paper, second-order statistics of IMI are studied in time-varying underwater acoustic particle velocity channels. First, the autocorrelation function, correlation coefficient, level crossing rate, and average outage duration of IMI are provided in a time-varying fading channel. Exact expressions are given in terms of the summation of special functions, which facilitate numerical calculations. Then, accurate approximations for the autocorrelation function and correlation coefficient are presented for low and high signal-to-noise ratios. Moreover, analytical and numerical results are provided for the correlation and level-crossing characteristics of IMI in underwater particle velocity channels. The results shed light on the dynamic behavior of mutual information in underwater acoustic particle velocity channels. [Work supported in part by the National Science Foundation (NSF), Grant CCF-0830190.]

4:15

1pUW13. Comparative sensitivity of pressure gradient receivers of force and inertial types to sound pressure in plane wave. Vladimir Korenbaum (Pacific Oceanologic Inst., 43, Baltiiskaya Str., Vladivostok 690041, Russian Federation, v-kor@poi.dvo.ru), Sergei Gorovoy (Far Eastern State Univ., Vladivostok, Russian Federation), Alexandr Tagiltcev, and Anatoly Kostiv (Pacific Oceanologic Inst., Vladivostok, Russian Federation)

Among known constructing schemes of pressure gradient receivers (PGR)—a difference type (including two-point version), a force type, and an inertial one—only the last two types are applicable when receiver size is much smaller than a wavelength of plane wave. Each of these two PGR types has certain advantages/disadvantages for various applications. The objective is a comparison of these PGRs, having identical external dimensions, by their sensitivity to sound pressure. In the long-wave two-dimensional approach the expression for a sensitivity to sound pressure is derived for a force type PGR, in which plate bending piezoelectric transducer is installed in a passage of the cylindrical housing. Also, the expression is developed for the sensitivity to sound pressure of an inertial type PGR with a one-component accelerometer embedded into its cylindrical housing. We derived the expression to estimate required one-side pressure sensitivity of the bending piezoelectric transducer of the force type PGR to equalize the sensitivities of PGRs of both types to sound pressure when the vibration sensitivity of the accelerometer is known. Possibilities of increasing sensitivity of the force type PGR are considered. [The study was supported by the grant 15-IV-1-001 of Far Eastern Branch of Russian Academy of Sciences.]

4:30

1pUW14. A wave structure based method for recognition of marine acoustic target signals. Qingxin Meng and Shie Yang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St.,Nangang District, Harbin City, Heilongjiang Province 150001, China, mengqingxin005@hrbeu.edu.cn)

The loudness and timbre of propeller are remarkable features of ship-radiated noise. The information of loudness and timbre is indicated in the wave structure of time series, and the feature of wave structure can be applied to classify various marine acoustic targets. In this paper, the feature extracting method of time series wave structure is studied. A nine-dimensional feature vector is constructed through statistical characteristics, containing zero-crossing wavelength, peek-to-peek amplitude, zero-crossing-wavelength difference and wave train areas. The feature vectors are inputted into SVM classifier to identify marine acoustic targets. The kernel function is set radial basis function (RBF).The penalty factor and kernel width of RBF are selected by the method of grid search. Finally, the recognition rate of test data reaches over 89.5%, with the help of cross validation. The sea-test data show the validity of target recognition ability of the method above.

4:45

1pUW15. Possibility of acoustic noise interferometry applications for passive remote sensing in shallow water. Sergei Sergeev, Andrey Shurup, Alisa Scherbina, and Pavel Mukhanov (Phys., Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, sergeev@aesc.msu.ru)

The cross-coherence function of the ambient noise field received during the quite long accumulation time by two hydrophones in the shallow water can exhibit two peaks. These peaks correspond to the travel times of signals between these two hydrophones. If the features of the shallow water are changed than the positions of these peaks are shifted. As result one can get information to reconstruct the physical properties and variability of the underwater environment using such peaks' shifts. In this work, the possibility of the passive remote sensing in shallow water features is considered based on the time-frequency analysis of the ambient noise cross-coherence function. The key point which define the possibilities of this method are the accumulation times required to estimate the peaks of the noise cross-coherence function with the acceptable signal-to-noise ratio. It is shown that the desired accumulation times can be considerably reduced by the appropriate choice of the frequency bands where the noise fields are formed by the small number of the incoherent hydroacoustic modes.

5:00

1pUW16. Eigenvector-based signal subspace estimation. Jorge E. Quijano (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr. A405, 3800 Finnerty Rd. (Ring Road), Victoria, British Columbia V8P 5C2, Canada, jorgeq@uvic.ca) and Lisa M. Zurk (Elec. and Comput. Eng. Dept., Portland State Univ., Portland, OR)

In this work, we explore the performance of a new algorithm for the estimation of signal and noise subspaces from limited data collected by a large-aperture sonar array. Based on statistical properties of scalar products between deterministic and complex random vectors, the proposed algorithm

defines a statistically justified threshold to identify target-related features (i.e., wavefronts) embedded in the sample eigenvectors. This leads to an improved estimator for the signal-bearing eigenspace that can be applied to known eigenspace beamforming processors. It is shown that data projection into the improved subspace allows better detection of closely spaced targets compared to current subspace beamformers, which utilize a subset of the unaltered sample eigenvectors for subspace estimation. In addition, the proposed threshold gives the user control over the maximum number of false detections by the beamformer. Simulated data are used to quantify the performance of the signal subspace estimator according to a normalized metric that compares estimated and true signal subspaces. Improvement on beamforming resolution using the proposed method is illustrated with simulated data corresponding to a horizontal line array, as well as experimental data from the Shallow Water Array Performance experiment.

5:15

1pUW17. Seismic sources in parabolic equation solutions for beach and island propagation scenarios. Scott D. Frank (Mathematics, Marist College, 3399 North Rd., Marist College, Poughkeepsie, NY 12601, scott.frank@marist.edu) and Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, Golden, CO)

Seismic events at and under the seafloor can be detected by seismometers on a beach. Events occurring in the earth's crust without ocean cover can also be detected at sea. Parabolic equation solutions for environments that include seismic sources in conjunction with island topography demonstrate that results for propagation moving away from or onto land can be obtained. Transmission loss results are given in the context of the (u_r,w) parabolic equation formulation of elasticity. Horizontal and vertical displacement values, as would be measured on land-based or ocean bottom seismometers, can also be obtained. Examples involving underwater volcanic events are considered, as are shore-bound events that generate oceanic *T*-waves that would be measured at sea.

MONDAY AFTERNOON, 18 MAY 2015 — DUQUESNE 1, 5:00 P.M. TO 6:00 P.M.

Meeting of Accredited Standards Committee (ASC) S2 Mechanical Vibration and Shock

C. F. Gaumond, Chair ASC S2
14809 Reserve Road, Accokeek, MD 20607

J. T. Nelson, Vice Chair ASC S2
Wilson Ihrig & Associates, Inc., 6001 Shellmound St., Suite 400, Emeryville, CA 94608

Accredited Standards Committee S2 on Mechanical Vibration and Shock. Working group chairs will report on the status of various shock and vibration standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 108, Mechanical vibration, shock, and condition monitoring, and four of its subcommittees, take note-that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 19 May 2015.

Scope of S2: Standards, specification, methods of measurement and test, and terminology in the field of mechanical vibration and shock, and condition monitoring and diagnostics of machines, including the effects of exposure to mechanical vibration and shock on humans, including those aspects which pertain to biological safety, tolerance, and comfort.

Session 1eED

Education in Acoustics and Women in Acoustics: Listen Up and Get Involved

Cameron T. Vongsawad, Cochair
Physics & Astronomy, Brigham Young University, 1041 E. Briar Avenue, Provo, UT 84604

Tracianne B. Neilsen, Cochair
Brigham Young University, N311 ESC, Provo, UT 84602

This workshop consists of a hands-on tutorial, interactive demonstrations, and a panel discussion about careers in acoustics. The primary goals of this workshop are to expose kids 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 email Traci Neilsen (tnb@byu.edu) if you have time to help with either guiding the kids 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.

Contributed Paper

5:30

1eED1. Visualization of multiple array acoustic source localization. Michael V. Scanlon (US Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783-1197, michael.v.scanlon2.civ@mail.mil)

Transient acoustic events such as gunfire, rocket launches, and explosions can be located using triangulation with multiple lines-of-bearing (LOBs) generated by distributed acoustic microphone arrays. The Army has fielded diverse sound localization and ranging systems that work well in open field environments, but suffer reduced performance in urban environments where echoes, multipath, diffraction, and blockages modify both the signatures and the propagation path. This demonstration will use light projected through a water basin to a screen to show traveling waves (ripple diffraction patterns) from transient events and how those wavefronts pass differently across the distributed array locations. The observers will gain an understanding of how multiple microphone elements in a small cluster of microphones (an array) can derive time-difference-of-arrival (TDOA) estimates to create array LOBs, as well as an introduction to triangulation through multiple distributed array LOBs. Objects simulating walls and buildings will be placed in the basin to demonstrate how complex waveforms and propagation paths can decrease the accuracy of acoustic localization systems.

Payment of separate registration fee required to attend.

MONDAY AFTERNOON, 18 MAY 2015 BALLROOM 3, 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Man-Made Noise and Aquatic Life: Data, Data Gaps, and Speculation

Micheal L. Dent, Chair
Psychology, University at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260

Invited Paper

7:00

1eID1. Man-made noise and aquatic Life: data, data gaps, and speculation. Arthur N. Popper (Dept. of Biology, Univ. of Maryland, 15501 Prince Frederick Way, Silver Spring, MD 20906, apopper@umd.edu)

This talk will consider how man-made sounds may impact aquatic life—with a focus on fishes (with possible digressions to invertebrates, marine mammals, and turtles). The talk will start with a discussion of a Jacques Cousteau movie and then ask why animals (and humans) hear. After considering how fishes (and invertebrates) hear, the focus of the talk will turn to potential impacts of sound on these animals and what is currently actually known (based on data and not speculation) about effects on physiology and behavior. The talk will conclude with a consideration of recent guidelines on effects of sounds and then provide an overview of data gaps and the most important areas for future research.

Session 2aAAa

Architectural Acoustics, Noise, Speech Communication, Psychological Physiological Acoustics, and Signal Processing in Acoustics: Sixty-Fifth Anniversary of Noise and Health: Session in Honor of Karl Kryter I

David M. Sykes, Cochair
Architectural Acoustics, Rensselaer Polytechnic Institute, 31 Baker Farm Rd., Lincoln, MA 01773

Michael J. Epstein, Cochair
Northeastern University, 360 Huntington Ave., 226FR, Boston, MA 02115

William J. Cavanaugh, Cochair
Cavanaugh Tocci Assoc. Inc., 3 Merifield Ln., Natick, MA 01760-5520

Chair's Introduction—8:00

Invited Papers

8:05

2aAAa1. Auditory and non-auditory effects of noise on health: An ICBEN perspective. Mathias Basner (Psychiatry, Univ. of Pennsylvania Perelman School of Medicine, 1019 Blockley Hall, 423 Guardian Dr., Philadelphia, PA 19104-6021, basner@upenn.edu)

Noise is pervasive in everyday life and induces both auditory and non-auditory health effects. Noise-induced hearing loss remains the most common occupational disease in the United States, but is also increasingly caused by social noise exposure (e.g., through music players). Simultaneously, evidence on the public health impact of non-auditory effects of environmental noise exposure is growing. According to the World Health Organization (WHO), ca. 1.6 Million healthy life years (DALYs) are lost annually in the western member states of the European Union due to exposure to environmental noise. The majority (>90%) of these effects can be attributed to noise-induced sleep disturbance and community annoyance, but noise may also interfere with communication and lead to cognitive impairment of children. Epidemiological studies increasingly support an association of long-term noise exposure with the incidence of hypertension and cardiovascular disease. Up-to-date exposure-response relationships are needed for health impact assessments, to reassess the validity of current noise policy and to better mitigate the negative health consequences of noise. The International Commission on Biological Effects of Noise (ICBEN) is a non-profit organization constituted 1978 that promotes all aspects of noise effects research and its application through International Noise Teams and an International Congress every three years.

8:35

2aAAa2. Karl Kryter and psychoacoustics at Bolt Beranek and Newman. Leo Beranek (none, none, 10 Longwood Dr., Westwood, MA 02090, Beranekleo@ieee.org)

Karl Kryter joined Bolt Beranek and Newman from Harvard in 1957. He was named the head of the Department of Psychoacoustics where he worked with his close friend, J. C. R. Licklider. His first work was with speech intelligibility in noise. He was the psychoacoustician on the team of three with Laymon Miller and Leo Beranekin a multi-year project for the Port York Authority that led to the reduction of noise created by the first jet commercial aircraft, the Boeing 707, by 15 dB. He was responsible for developing a means for measuring the perception by people of the loudness of noise—particularly aircraft noise. Named the perceived noise decibel PNDB, it remains today the accepted method for measuring the effect of noise on people living around airports. After that he did sleep research until he departed in 1966 for the Stanford Research Institute.

8:55

2aAAa3. Effects of hospital noise on speech intelligibility. Frederick J. Gallun (VA Portland Health Care System, National Ctr. for Rehabilitative Auditory Res., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov)

The most straightforward way to consider the effects of noise on the ability to understand speech is in terms of overlap of energy at the level of the cochlea. This approach, sometimes called "energetic masking" has been well studied and is quantified in a number of international standards such as the Speech Intelligibility Index (ANSI, 2004). Recently, however, another aspect of noise has captured the interest of those studying speech intelligibility. Known as "informational masking," this refers to the ability of noise to interfere with intelligibility even when energetic overlap is low. A study describing this phenomenon with respect to hospital noise will be described and it will be demonstrated that the impact of human voices on speech intelligibility provides a difficult problem for those trying to predict and quantify the effects of noise based only on the statistics of the acoustical signal.

9:15

2aAAa4. A tribute to Karl Kryter. Karl S. Pearsons (22689 Mulholland Dr., Woodland Hills, CA 91364-4941, karlkity@pacbell.net)

Karl Kryter came to Bolt Beranek and Newman in 1957 from Harvard to lead the Psycho-Acoustics Department. At that time, commercial aviation was transitioning from propeller driven aircraft to jet powered aircraft, like the Douglas DC8 and the Boeing 707. Public concern about the perceived increase in noise prompted the development of noise metrics. These new metrics provided better correlation with judged annoyance than overall sound levels; thus, perceived noise level (PNL) in PNdB units was introduced as a measure of airplane noise. He also contributed to establishing noise levels which may cause permanent hearing loss. Another interest was in creating a device that masked the noise of a dentist's drill. It was called the audio-analgesic. The last time I saw Karl was when we represented opposite sides in a court case. It was about the noise level of a medivac helicopter landing on a hospital roof. He was a wonderful boss and friend, a true role model. Fellow colleagues had only praise for him and the work environment he created. Born in 1914, he almost made it to 100 years, passing in 2013.

9:35

2aAAa5. Prescribing healthy hospital soundscapes. Erica E. Ryherd (Durham School of Architectural Eng. & Construction, Univ. of Nebraska - Lincoln, Omaha, NE, eryherd@unl.edu) and Kerstin Persson Waye (Occupational & Environ. Medicine, The Sahlgrenska Acad., Gothenburg Univ., Gothenburg, Sweden)

In *The Effects of Noise on Man*, Karl Kryter emphasized that human reaction to sound is quantitatively related to the physical nature of sounds. He specified five unwanted characteristics of sound: "(a) the masking of unwanted sounds, particularly speech, (b) auditory fatigue and damage to hearing, (c) excessive loudness, (d) some general quality of bothersomeness or noisiness, and (e) startle." Hospital soundscapes have been shown to demonstrate all five characteristics in one fashion or another. Some unwanted effects of sound have also been shown, such as fragmented sleep and recuperation, cardiovascular response, pain and intensive care delirium—however, few studies have been able to causally link sounds to patient outcomes and few examine detailed characteristics other than loudness. This paper will summarize what we know about noise in hospitals and the health effects on occupants, including highlights from the Healthcare Acoustics Research Team (HART) body of research. Results will be used to identify important areas for future research.

9:55–10:05 Break

10:05

2aAAa6. *Zeitgeist* from Kryter's work at the PsychoAcoustics Laboratory at Harvard to the present. Mary Florentine (Dept. of Commun. Sci. and Disord. (226-FR), Dept. of Elec. and Comput. Eng., Northeastern Univ., Bldg. 226 FR, 360 Huntington Ave., Boston, MA 02115, m.florentine@neu.edu) and Michael Epstein (Auditory Modeling and Processing Lab., Dept. of Commun. Sci. and Disord., Dept. of Elec. and Comput. Eng., Dept. of BioEng., Northeastern Univ., Boston, MA)

Kryter's writings indicate that he was keenly aware that intense sounds damage auditory systems. He was also aware that there is no one-to-one correspondence between the physical world and our perception of it. In fact, much of Kryter's work sought to summarize that relationship in general terms as they applied to average observers. He consistently sought an inclusive theoretical framework. Kryter would have been fascinated by the recent discoveries regarding individual differences among listeners with different types of hearing losses. He would have delighted in the recent connections made between psychoacoustics and physiology, and the trend toward ecologically valid research. This presentation traces the *zeitgeist* from his work at the PsychoAcoustics Laboratory (PAL) at Harvard to the present, examining loudness across the entire dynamic range for listeners with normal hearing and different types of hearing losses. [This work was supported, in part, by NIH-NIDCD grants 1R03DC009071 and R01DC02241.]

10:25

2aAAa7. Neurophysiological effects of noise-induced hearing loss. Michael G. Heinz (Speech, Lang., and Hearing Sci. & Biomedical Eng., Purdue Univ., 715 Clinic Dr., West Lafayette, IN 47907, mheinz@purdue.edu)

Noise over-exposure is known to cause sensorineural hearing loss (SNHL) through damage to sensory hair cells and subsequent cochlear neuronal loss. These same histopathological changes can result from other etiological factors (e.g., aging, ototoxic drugs). Audiologically, each is currently classified as SNHL, despite differences in etiology. Functionally, noise-induced hearing loss can differ significantly from other SNHL etiologies; however, these differences remain hidden from current audiological diagnostic tests. For example, noise-induced mechanical damage to stereocilia on inner- and outer-hair cells results in neurophysiological responses that complicate the common view of loudness recruitment. Also, mechanical effects of noise overexposure appear to affect the tip-to-tail ratio of impaired auditory-nerve-fiber tuning curves in ways that create a much more significant degradation of the cochlear tonotopic representation than does a metabolic form of age-related hearing loss. Even temporary threshold shifts due to moderate noise exposure can cause permanent synaptic loss at inner-hair-cell/auditory-nerve-fiber synapses, which reduces neural-coding redundancy for complex sounds and likely degrades listening in background noise. These pathophysiological results suggest that novel diagnostic measures are now required to dissect the single audiological category of SNHL to allow for better individual fitting of hearing aids to restore speech intelligibility in real-world environments. [Work supported by NIH.]

10:45

2aAAa8. Noise-induced cochlear synaptopathy: Extending effects of noise on man? Sharon G. Kujawa (Massachusetts Eye and Ear Infirmary and Harvard Med. School, 243 Charles St., Boston, MA 02114, sharon_kujawa@meei.harvard.edu)

Kryter wrote that "*…methods commonly used in medicine for the evaluation of impairment to hearing and the relation of this impairment to noise exposure may lead to significant underestimates of the severity of noise-induced hearing impairment and overestimations of tolerable limits for exposure to noise*." (J. Acoust. Soc. Am. **53**(5), 1211–1234). Although specifics differ, recent work in our laboratory provides strong evidence supporting this assertion. In a series of investigations conducted over the last 5 years, we have

shown that a primary consequence of noise exposure is acute loss of synapses between cochlear inner hair cells (IHC) and auditory nerve fibers followed by loss of the affected neurons themselves. Losses are robust, permanent, and progressive, even for temporary threshold shift-producing exposures, and aging of these ears is exaggerated, going forward. This noise-induced injury can hide behind normal thresholds, our traditional evidence of recovery; however, communication between IHCs and nerve fibers is nevertheless permanently interrupted. We hypothesize that these synaptopathic and neurodegenerative consequences of noise may contribute to speech discrimination difficulties in challenging listening environments. This talk will summarize what we know about structural and functional consequences and noise-risk implications of noise-induced cochlear synaptopathy. [Research supported by R01 DC 008577.]

11:05

2aAAa9. Age-related hearing loss, noise-induced hearing loss, and speech-understanding performance. Larry E. Humes (Dept. Speech & Hearing Sci., Indiana Univ., Bloomington, IN 47405-7002, humes@indiana.edu)

Karl Kryter was keenly interested in the effects of noise on hearing, the interaction of age-related hearing loss (ARHL) with noise-induced hearing loss (NIHL), and, ultimately, the consequences of both on an individual's ability to understand speech in quiet and in noise. This review will touch on each of these topics and the interrelationships among them. The focus of this presentation will be on discussion of factors impacting the speech-understanding performance of older adults, with parallels drawn for NIHL where possible. Factors considered will include those represented by a variety of peripheral-auditory, central-auditory, and cognitive measures. Interactions among these factors will also be reviewed.

Contributed Paper

11:25

2aAAa10. The effects of noise on physician cognitive performance in a hospital emergency department. Peter Dodds, Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 161 Washington St. #4, Troy, NY 12180, phdodds@gmail.com), David Sykes (Acoust. Res. Council, Lincoln, MA), Wayne Triner, and Linda Sinclair (Albany Medical Ctr., Albany, NY)

The contemporary hospital emergency department presents a noisy and distracting sonic environment. While previous research has shown that the highly variable and overloud soundscape of hospitals leads to increased annoyance and physical stress for hospital employees, there is a need to objectively quantify the cognitive effects of noise on healthcare providers, particularly physicians. An ongoing collaborative research effort by the authors seeks to better understand and measure the cognitive and psychoacoustic effects of noise in a busy urban emergency department. Using binaural room-acoustic measurements and psychoacoustic modeling, the soundscape of the emergency department at Albany Medical Center has been analyzed. Through the use of the n-Back cognitive test and calibrated binaural recordings and room-simulations, we are examining the effect of a variety of sonic phenomena on the working memory and cognitive control of emergency department physicians at the Albany Medical Center at various points throughout their shifts. This paper will discuss methods for in situ cognitive testing, preliminary results, and review possible interventions and their efficacy in reducing noise and distractions for physicians in the emergency department.

11:40–12:00

Panel Discussion and Open Microphone

TUESDAY MORNING, 19 MAY 2015 BALLROOM 2, 9:30 A.M. TO 12:30 P.M.

Session 2aAAb

Architectural Acoustics and National Council of Acoustical Consultants: Student Design Competition

David S. Woolworth, Cochair
Oxford Acoustics, 356 CR 102, Oxford, MS 38655

Andrew N. Miller, Cochair
Bai, LLC

Norman H. Philipp, Cochair
Geiler and Associates, LLC, 1840 E. 153rd. Cir., Olathe, KS 66062

The Technical Committee on Architectural Acoustics of the Acoustical Society of America with support from the Robert Newman Student Award Fund and the National Council of Acoustical Consultants are sponsoring the 2015 Student Design Competition that will be professionally judged at this meeting. The competition involves the design of a performance venue in addition to a casino and hotel facility in downtown Pittsburgh, Pennsylvania. The submitted designs will be judged by a panel of professional architects and acoustical consultants. An award of US$1250 will be made to the submitter(s) of the design judged "first honors." Four awards of US$700 each will be made to the submitters of four entries judged "commendation."

Session 2aAB

Animal Bioacoustics and Psychological and Physiological Acoustics: Auditory Scene Analysis in Natural Environments

Annemarie Surlykke, Cochair
Biology, University of Southern Denmark, Campusvej 55, Odense DK-5230, Denmark

Susan Denham, Cochair
School of Psychology, University of Plymouth, Plymouth pl4 8aa, United Kingdom

Invited Papers

8:00

2aAB1. Computational issues in natural auditory scene analysis. Michael S. Lewicki (Dept. of Elec. Eng. and Comput. Sci., Case Western Reserve Univ., CWRU/EECS Glennan Rm. 321, 10900 Euclid Ave., Cleveland, OH 44106, michael.lewicki@case.edu), Bruno A. Olshausen (Redwood Ctr. for Theor. Neurosci., UC Berkeley, Berkeley, CA), Annemarie Surlykke (Dept. of Biology, Univ. of Southern Denmark, Odense, Denmark), and Cynthia F. Moss (Dept. of Psychol. & Brain Sci., Johns Hopkins Univ., Baltimore, MD)

Scene analysis is a complex process involving a hierarchy of computational problems ranging from sensory representation to feature extraction to active perception. It is studied in a wide range of fields using different approaches, but we still have only limited insight into the computations used by biological systems. Experimental approaches often implicitly assume a feature detection paradigm without addressing the true complexity of the problem. Computational approaches are now capable of solving complex scene analysis problems, but these often defined in a way that is of limited relevance to biology. The challenge is to develop approaches for deducing computational principles relevant to biological systems. I will present the view that scene analysis is a universal problem solved by all animals, and that we can gain new insights by studying the problems that animals face in complex natural environments. From this, I will present framework for studying scene analysis comprising four essential properties: (1) the ability to solve ill-posed problems, (2) the ability to integrate and store information across time and modality, (3) efficient recovery and representation of 3D scene structure, and (4) the use of optimal motor actions for acquiring information to progress toward behavioral goals.

8:20

2aAB2. The role of form in modeling auditory scene analysis. Susan Denham and Martin Coath (School of Psych., Univ. ersity of Plymouth, Plymouth pl4 8aa, United Kingdom, s.denham@plymouth.ac.uk)

Separating out and correctly grouping the sounds of a communicating animal from the natural acoustic environment poses significant challenges to models of auditory scene analysis, yet animals perform this task very effectively. To date, most models have focussed on simultaneous grouping cues and the segregation of discrete sound events, although some take the longer term context into account. Inspired by the important part that form plays in the segregation and recognition of visual objects, we consider the role of form in auditory scene analysis. By form in audition we mean the dynamic spectrotemporal patterns characterizing individual sound events as well as their timing with respect to each other. We present a model capable of segregating and recognizing natural communication calls within complex acoustic environments. Incoming sounds are processed using a model of the auditory periphery and fed into a recurrent neural network that rapidly tunes itself to respond preferentially to specific events. Representations of predictable patterns of events in the sequence are created on the fly and maintained on the basis of their predictive success and conflict with other representations. Activation levels of these representations are interpreted in terms of object recognition.

8:40

2aAB3. Audio-vocal feedback in bats and new roles for echolocation calls in social communication. Kirsten M. Bohn (Florida Int.. Univ., 518 Giralda Ave., Coral Gables, FL 33134, bohnkirsten@gmail.com) and Michael Smotherman (Texas A&M Univ., College Station, TX)

An important aspect of auditory scene analysis is the specialized neurocircuitry required for vocal production. in a dynamic acoustic environment. Although often taken for granted, enhanced audio-vocal feedback is relatively uncommon in animals and yet an important precursor to vocal learning. We argue that vocal complexity in bats is an exaptation of the highly specialized audio-vocal feedback system that has evolved for echolocation. First, we explore how audio-vocal feedback enhances echolocation. Second, we review how echolocation pulses serve social functions by providing information to receivers (like gender, identity, or food availability). Third, using our research on molossid bats (family Molossidae), we explore whether vocal plasticity in sonar has contributed to an expanded role for sonar pulses in social communication. In at least three molossids, roosting bats rapidly sing in response to echolocation pulses of flying conspecifics. However, more importantly, we show that in multiple species, echolocation is produced in purely social contexts. Roosting bats embed pulses and feeding buzzes into their courtship songs that are not acoustically distinct than when foraging. Finally, some

molossids not only sing in roosts, but also in flight—that is, they echolocate and sing simultaneously. These findings indicate that echolocation plays even more of a role in social communication than commonly believed and that the production of echolocation and social communication is tightly coupled and coordinated at a high level.

9:00

2aAB4. Are the mechanisms for stream segregation shared among anurans and other tetrapods? Jakob Christensen-Dalsgaard (Biology, Univ. of Southern Denmark, Campusvej 55, Odense M DK-5230, Denmark, jcd@biology.sdu.dk)

Many male anurans (frogs and toads) call in large aggregations. Since the fundamental task of anuran auditory communication probably is to attract and localize potential mates, segregation of the callers is likely an important task for the auditory system. Behavioral experiments have shown that elements of stream segregation, based on frequency separation and spatial separation, can be demonstrated in anurans. The neural processing of these cues is interesting, because most auditory processing probably takes place in the midbrain torus semicircularis (TS, homolog to the inferior colliculus). It has been shown that spatial release from masking is sharpened by 6 dB in the TS, and that neurons in the TS are selective for call rates and number of call pulses. However, recently electrical stimulation of thalamic structures have demonstrated possible attentional modulation of TS responses that could also function in stream segregation. The modulation was call-specific and could also enhance binaural cues. In conclusion, many of the elements involved in auditory segregation in other animals are also found in anurans. However, it is uncertain whether the underlying mechanisms are similar. One crucial element in primate sound segregation—the representation of the competing streams in the neural responses—has so far not been demonstrated in anurans, and the most robust elements of stream segregation is really based on frequency processing, a relatively simple and ubiquitous property of most auditory systems.

9:20–9:35 Break

9:35

2aAB5. Measurement and control of vocal interactions in songbirds. Richard Hahnloser (Inst. of Neuroinformatics, ETH Zurich and Univ. of Zurich, Winterthurerstrasse 190, Zurich, Schweiz 8057, Switzerland, rich@ini.phys.ethz.ch)

One obstacle for investigating vocal interactions in vertebrate animals is difficulty of discriminating individual vocalizations of rapidly moving, sometimes simultaneously vocalizing individuals. To overcome this obstacle, we have developed an ultra-miniature back-attached sound/acceleration recording system that allows perfect separation of birdsong vocalizations irrespective of background noise and the number of vocalizing animals nearby. With this system, we have been able to identify hierarchies in the vocal interactions among adult male and female zebra finches. Furthermore, to causally interfere with vocal interactions in groups of up to four birds, we have developed an echo cancelation system that allows us to have full digital control of vocal interactions between any bird pair. With these tools, we hope to be able to further advance the dissection of vocal communication in songbirds.

9:55

2aAB6. Listening through the ears of echolocating *Myotis daubentonii* bats hunting in groups. Annemarie Surlykke, Mads N. Olsen (Biology, Univ. of Southern Denmark, Campusvej 55, Odense DK-5230, Denmark, ams@biology.sdu.dk), and Carien Mol (Biology, Univ. of Southern Denmark, Utrecht, Netherlands)

Echolocation allows bats to orient and catch insects in the dark. One intriguing question is how bats are able to recognize their own echoes when hunting in groups. Bats can adjust their call frequency, but this strategy may not be efficient for species like *Myotis daubentonii* emitting broadband frequency modulated signals. However, the actual masking may be reduced for bats like *M. daubentonii* emitting short directional signals with low duty cycle. We used a 12-microphone array and infrared camera to record flight and vocal behavior of groups of Daubenton's bat hunting over a river in the field. We used flight path reconstructions to analyze the acoustic world from the bat's perspective. For the focal bat, we reconstructed (1) its own emissions, (2) the pulses from conspecifics nearby, (3) its own insect echoes, and (4) insect echoes from pulses of the other bat. The data showed that when two bats fly together echoes were only rarely overlapped by the other sounds. We here provide a framework for obtaining detailed information of flight and echolocation behavior in complex natural surroundings, emphasizing the importance of adapting a "bat's viewpoint" when studying natural echolocation.

Contributed Papers

10:15

2aAB7. Nocturnal peace at a Conservation Center for Species Survival? Susan M. Wiseman (None, Waco, TX) and Preston S. Wilson (Mech. Eng., The Univ. of Texas at Austin, 204 East Dean Keeton St., Austin, TX 78712, pswilson@mail.utexas.edu)

E.O. Wilson suggested that the natural world is the most information-rich acoustic environment, especially for animals and indigenous peoples, and that is directly and indirectly essential to their survival (Wilson, 1984, *Biophilia*, Harvard University Press). Agriculturalists still value and constantly monitor their soundscape since it provides invaluable cues that may lead to success or failure. Krause reports that healthy natural soundscapes comprise a myriad of biophony, and indeed the ecological health of a region can be measured by its diverse voices (Krause, 1987, "Bio-acoustics: Habitat ambience & ecological balance," *Whole Earth Rev*.). But do such soundscapes fall silent for much of the night? And are there extensive periods of silence at a highly successful Conservation Center for Species Survival? This study analyzes the soundscape continuously recorded at Fossil Rim Wildlife Center in Texas for a week during Fall 2013, to determine the prevalence of quiet periods and the acoustic environment in which such periods appeared to occur.

10:30

2aAB8. Active enhancement found in responses of the cochlear nerve to detect weak echoes created by the auditory periphery in Japanese house bat, *Pipistrelus abramus*. Hiroshi Riquimaroux, Hiroshi Onodera (Life and Medical Sci., Doshisha Univ., 1-3 Miyakotani, Tatara, Kyotanabe 610-0321, Japan, hrikimar@mail.doshisha.ac.jp), and Ikuo Matsuo (Tohoku-gakuin Univ., Sendai, Japan)

Characteristics of frequency tunings related to echolocation were investigated in an FM bat, Japanese house bat (*Pipistrellus* abramus). A particular attention was paid on their terminal frequency of FM sweep. When search for preys in the field, they stretch duration of echolocation pulses and reduce FM sweeping range, resulting in pseudo constant frequency pulses. Distribution of best frequency in the neurons in the inferior colliculus indicated that about 50% of neurons appeared to be tuned to the pseudo constant frequency, around 40 kHz. The compound action potentials were recorded from medulla oblongata through a metal telectrode. We hypothesized that the bat has a peripheral system to enhance frequency component of 40 kHz to be detected easier than other frequency components. Active enhancement of 40 kHz frequency component by the outer hair cell and overwhelming number of cochlear nerves in charge of 40 kHz would create low threshold and high resolution sensitivity for frequency range around 40 kHz. Results of the present study show that compound action potential of the cochlear nerve (N1) generates nonlinearly enhanced amplification in frequency range of 40 kHz but linear amplification in lower frequency ranges.

10:45

2aAB9. Evaluation of a simple model for the acoustics of bat swarms. Mingyi Liu and Rolf Müller (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, lmingyi@vt.edu)

Bats using their biosonar while flying in dense swarms may face significant bioacoustic challenges, in particular mutual sonar jamming. While possible solutions to the jamming problem have been investigated several times, the severity of this problem has received far less attention. To characterize the acoustics of bat swarms, a simple model of the acoustically relevant properties of a bat swarm has been evaluated. The model contains only four parameters: bat density, biosonar beamwidth, duty cycle, and a scalar measure for the smoothness in the flight trajectories. In addition, a threshold to define substantial jamming was set relative to the emission level. The simulations results show that all four model parameters can have a major impact on jamming probability. Depending on the combination of parameter values, situations with or without substantial jamming probabilities could be produced within reasonable ranges of all model parameters. Hence, the model suggests that not every bat swarm does necessarily impose grave jamming problems. Since the model parameters should be comparatively easy to estimate for actual bat swarms, the simulation results could give researchers a way to assess the acoustic environment of actual bat swarms and determine cases where a study of biosonar jamming could be worthwhile.

11:00

2aAB10. Noise induced threshold shifts after noise exposure: Are bats special? Andrea Simmons (Brown Univ., Box 1821, Providence, RI 02912, Andrea_Simmons@brown.edu), Michaela Warnecke (Johns Hopkins Univ., Baltimore, MD), Kelsey Hom, and James Simmons (Brown Univ., Providence, RI)

We conducted psychophysical and neurophysiological experiments to test the hypothesis that big brown bats suffer less severe temporary threshold shifts after noise exposure than other small mammals that also hear high frequency sounds. Five big brown bats were trained in psychophysical detection experiments to obtain thresholds to FM sweeps spanning the frequency range of their echolocation calls. Bats were then exposed to 115 dB of broadband noise for one hour, and thresholds re-measured 20 min and 24 hour after exposure. For all bats, threshold elevations at 20 min post-exposure were 3 dB or less, while at 24 hour post-exposure, thresholds were similar to those obtained pre-exposure. Local field potentials were recorded in the cochlear nucleus of anesthetized big brown bats before and after noise exposure. Neural thresholds to FM sweeps and to single tone frequencies were unaffected by noise exposure. These data suggest that big brown bats are an excellent model system to study naturally occurring immunity to noise damage. [Work supported by ONR and the Capita Foundation.]

11:15

2aAB11. Underwater auditory scenes near coral reefs. Erica Staaterman and Claire Paris (Appl. Marine Phys. & Marine Biology and Fisheries, Univ. of Miami, 4600 Rickenbacker Cswy., Miami, FL 33149, e.staaterman@gmail.com)

Recent studies suggest that reef soundscapes may serve as long-distance navigational cues for settlement-stage larval fishes. To understand the role of acoustic signals during larval settlement, we investigated temporal and spatial patterns in coral reef soundscapes in the Florida Keys, USA. We used 14-month simultaneous acoustic recordings from two nearby reefs, coupled with environmental data, to describe temporal variability in the soundscape on scales of hours to months, and to understand abiotic and biological components. We also recorded acoustic pressure and particle acceleration (which fish larvae likely perceive) with increasing distance from the reef. High acoustic frequencies typically varied on daily cycles, while low frequencies were primarily driven by lunar cycles. Some of these patterns were explained by environmental conditions, while others were attributed to biological sounds. At both reefs, the highest sound levels (~130 dB re:1μPa) occurred during new moons of the wet season, when many larval organisms settle on the reefs. Acoustic particle acceleration did not propagate as far as predicted from the plane-wave equation. The patterns uncovered here provide valuable insights into underwater acoustic scenes, and this study represents an example of novel acoustic instruments and analysis techniques that can be applied to any ecosystem.

Session 2aBA

Biomedical Acoustics: Ultrasound Contrast Agents: Nonlinear Bubble Dynamics

Tyrone M. Porter, Cochair
Boston University, 110 Cummington Mall, Boston, MA 02215

Jonathan A. Kopechek, Cochair
University of Pittsburgh, 200 Lothrop Street, Pittsburgh, PA 15213

Invited Papers

8:00

2aBA1. Role of lipid coating properties on nonlinear microbubble dynamics. Klazina Kooiman (Dept. of Biomedical Eng., Thorax Ctr., Erasmus MC, P.O. Box 2040, Rm. Ee2302, Rotterdam 3000 CA, Netherlands, k.kooiman@erasmusmc.nl)

Although nonlinear microbubble dynamics are exploited in many contrast-enhanced ultrasound imaging modalities, the role of lipid coating properties on nonlinear dynamics is not fully understood. Our studies show that microbubble vibrations differ between bubbles with homogeneous lipid distribution (main component DPPC, C16) and bubbles with heterogeneous lipid distribution (main component DSPC, C18). Relative to DPPC microbubbles, DSPC microbubbles were shown to elicit more second harmonic emissions, but exhibit a lower propensity to initiate subharmonic emissions. This suggests that homogeneous microstructures favor subharmonic emissions, whereas heterogeneous microstructures produce higher second harmonic emissions. Additionally, the nonlinear "compression-only" oscillation behavior is hypothesized and theoretically shown to relate to buckling of the lipid coating. High-speed fluorescence recordings revealed for the first time formation of hot spots (i.e., high local concentrations of lipids) in the coating of oscillating microbubbles during the compression phase, suggesting evidence of buckling/folding of the coating. However, hot spots were not correlated to compression-only behavior, but were related to the level of compression of the microbubbles during insonification, with a relative compression threshold of 15–20%. This suggests that compression-only behavior is not directly related to buckling of the lipid coating. This work sheds insight into the role of lipid coating properties on nonlinear microbubble behavior.

8:30

2aBA2. Ultrafast frame rate microscopy of microbubble oscillations: Current studies employing the UPMC-Cam. Brandon Helfield, Xucai Chen, Bin Qin, and Flordeliza Villanueva (Ctr. for Ultrasound Molecular Imaging and Therapeutics, Heart and Vascular Inst., Univ. of Pittsburgh Medical Ctr., 966 Scaife Hall Fl. 3550 Terrace St., Pittsburgh, PA 15213, helfieldb@upmc.edu)

Ultrasound-stimulated microbubbles are clinically employed for diagnostic imaging and have been shown to be a feasible therapeutic strategy for localized drug and gene delivery applications. In order to investigate microbubble oscillation dynamics, the University of Pittsburgh Medical Center has developed the UPMC-Cam—an ultrafast imaging system capable of recording 128 frames at up to 25 million frames per second, one of only two in the world currently in use. Current studies include elucidating the effect of fluid viscosity on microbubble behavior. At lower pressures (0.1–0.25 MPa), an increase in fluid viscosity from 1 to 4 cP alters the fundamental and second-harmonic oscillation amplitudes in a bubble size-dependent manner, related to resonance. At higher pressures (0.5–1.5 MPa), a significant decrease in the propensity for microbubble fragmentation was observed in the more viscid fluid environment. Additionally, studies aimed at gaining physical insights into sonoporation have been conducted in which individual bubbles lay adjacent to a cell monolayer. These studies have revealed a maximum microbubble expansion threshold above which sonoporation occurs, likely related to the associated shear stresses exerted by a microbubble on the adjacent cell membrane. The UPMC-Cam has shown to be an invaluable tool for investigating biophysical-related phenomena over a wide range of ultrasound-microbubble applications.

Contributed Papers

9:00

2aBA3. Radial excursions of bound and non-bound targeted lipid-coated single microbubbles. Tom van Rooij, Antonius F. van der Steen, Nico de Jong, and Klazina Kooiman (Dept. of Biomedical Eng., Thorax Ctr., Erasmus MC, Postbus 2040, Rotterdam 3000 CA, Netherlands, t.vanrooij@erasmusmc.nl)

One of the main challenges for ultrasound molecular imaging is acoustically distinguishing non-bound microbubbles from those that have bound to their molecular target. We previously showed that biotinylated DPPC-based microbubbles (16 C-atoms) had a larger binding area and a more domed shape when bound to a streptavidin-coated surface than DSPC-based microbubbles (18 C-atoms) [1]. In the present *in vitro* study, we used the Brandaris 128 ultrahigh-speed camera (~15 Mfps) to compare the acoustical responses of biotinylated DPPC and DSPC-based microbubbles in a non-bound configuration and bound to a streptavidin-coated membrane, aiming to acoustically discriminate them from each other. The microbubbles were driven at a pressure of 50 kPa and at frequencies between 1 and 4 MHz. The main difference between bound and non-bound microbubbles was the lower radial excursion at the fundamental frequency for bound microbubbles.

Resonance frequencies and subharmonic responses were the same for bound and non-bound microbubbles. Finally, at the second harmonic frequency, we found higher relative radial excursions for bound DSPC-based microbubbles than for non-bound DSPC microbubbles, whilst there was no difference for DPPC-based microbubbles. This might provide opportunities to acoustically discriminate bound from non-bound DSPC microbubbles. [1] Kooiman *et al*., Eur. J. Lipid Sci. Technol. (2014).

9:15

2aBA4. Heat and mass transfer effects on forced radial oscillations in soft tissue. Carlos Barajas and Eric Johnsen (Mech. Eng., Univ. of Michigan, 500 South State St., Ann Arbor, MI 48109, carlobar@umich.edu)

Cavitation has a vast array of biomedical purposes such as histotripsy, used to mechanically fractionate soft tissue. Alternatively, cavitation can cause unwanted biological damage in diagnostic ultrasound, e.g., when using ultrasound contrast agents. These benefits and harmful effects make it desirable to develop models to better understand cavitation in viscoelastic media. We numerically model cavitation in a viscoelastic (Kelvin-Voigt) medium using the Keller-Miksis equation to describe the radial bubble motion. We specifically focus on the effects of heat and mass transfer, through numerical solution of the full energy and mass equations, reduced models, and theoretical analysis. We will discuss how thresholds on the driving pressure and frequency where thermal and mass transfer effects become significant can be determined, as well as ranges of amplitudes for which reduced models can be used accurately. We will also present theoretical investigations on how oscillation properties depend on these effects. Our findings will provide guidelines to determine regimes when heat and mass transfer are dominant. [This work was supported in part by NSF grant number CBET 1253157 and NIH grant number 1R01HL110990-01A1.]

9:30

2aBA5. An *in vitro* study of subharmonic emissions from monodisperse lipid-coated microbubbles. Qian Li (Biomedical Eng., Boston Univ., 44 Cummington Mall, Rm. B01, Boston, MA 02215, qianli@bu.edu), Mark Burgess, and Tyrone Porter (Mech. Eng., Boston Univ., Boston, MA)

Subharmonic imaging is an ultrasound imaging method which utilizes the subharmonic response of a contrast agent to ultrasound excitation. It is possible to achieve high agent-to-tissue contrast because tissues do not emit subharmonic signals. In this project, we investigated the relationship between subharmonic emissions from monodisperse lipid-coated microbubbles and acoustic pressure. First, the resonance frequency for monodisperse microbubble suspension was determined from attenuation spectra measured using a through-transmission technique. Next, the microbubbles were excited at the resonance and at twice the resonance frequency. A transducer positioned orthogonal to the excitation transducer was used to detect acoustic emissions from the microbubbles at half the excitation frequency. It was found that the pressure required for subharmonic emissions was lower when monodisperse microbubbles were excited at twice the resonance frequency. For example, 6.5μm diameter microbubbles with a resonance frequency of 1.4 MHz had a pressure threshold for subharmonic emissions of approximately 30 kPa at 2.8 MHz excitation while those excited at 1.4 MHz could not be forced into subharmonic oscillations for the pressure range we used in this study (i.e., below 150 kPa). Implications of these results on the use of monodisperse lipid-coated microbubbles for subharmonic imaging will be discussed.

9:45

2aBA6. Exploitation of nonlinear acoustical effects of air bubbles in water for a bubble/target discriminating sonar. Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, klee@arlut.utexas.edu) and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Target detection by traditional sonar can be undermined by air bubbles in various environments, including the surf zone, within gas-bearing sediments, or within seagrass beds. Bubbles introduce a variety of strong acoustical effects, including the onset of many nonlinear effects at sound pressure levels far below that required in bubble-free water. This paper describes laboratory tank experiments demonstrating a bubble/target discriminating sonar that exploits some of these nonlinear effects. Bubble plumes were generated in the tank by releasing compressed air through a porous ceramic plate. Rigid targets were also positioned at various locations in the tank. A high amplitude driving pulse excited translation of bubbles in the water column yielding Doppler signatures from conventional imaging sonar, while Doppler signatures were absent from the heavy rigid targets ensonified by the same driving signal. For sufficient bubble density, subharmonic generation was also observed in regions of bubbly water, but not from the rigid targets. The goal of this work is to use one or both of these nonlinear effects to identify and reduce clutter caused by bubbles and improve target detection for sonar operation in areas where bubbles are present. [Work supported by ARL:UT and ONR.]

10:00–10:15 Break

10:15

2aBA7. Singular value decomposition: A better way to analyze cavitation-noise data. Parag V. Chitnis (Dept. of BioEng., George Mason Univ., 4400 University Dr., 1G5, Fairfax, VA 22032, pchitnis@gmu.edu), Caleb Farny (Dept. of Mech. Eng., Boston Univ., Boston, MA), and Ronald A. Roy (Dept. of Mech. Eng., Univ. of Oxford, Oxford, United Kingdom)

Detection of inertial and stable cavitation is important for guiding high-intensity focused ultrasound (HIFU). Acoustic transducers can passively detect broadband noise from inertial cavitation and the scattering of HIFU harmonics from stable cavitation bubbles. Conventional approaches to separating these signals typically involve a custom comb-filter applied in the frequency domain followed by an inverse-Fourier transform, which cannot be implemented in real-time. We present an alternative technique based on singular value decomposition (SVD) that efficiently separates the broadband emissions and HIFU harmonics in a single step. Spatio-temporally resolved cavitation detection was achieved using a 128-element, 5-MHz linear-array system operating at 15 frames/s. A 1.1-MHz transducer delivered HIFU to tissue-mimicking phantoms for a duration of 5 s. Beamformed radiofrequency signal corresponding to each scan line and frame were assembled into a matrix and SVD was performed. Eigen vectors that corresponded to HIFU harmonics were identified as ones whose spectrum contained a peak centered at one of the harmonic frequencies. The projection of data onto this eigen-base produced the stable-cavitation signal. The remaining eigenvectors were used to obtain the inertial-cavitation signal. The SVD-based method faithfully reproduced the structural details in the spatio-temporal cavitation maps produced using the comb-filter albeit with marginally lower SNR.

10:30

2aBA8. Microbubble behavior during long tone-burst ultrasound excitation. Xucai Chen, Jianjun Wang, John Pacella, and Flordeliza S. Villanueva (Ctr. for Ultrasound Molecular Imaging and Therapeutics, Univ. of Pittsburgh Medical Ctr., 3550 Terrace St., Scaife Hall, Rm. 969, Pittsburgh, PA 15261, chenxx2@upmc.edu)

Ultrasound-microbubble mediated therapies have been investigated to restore perfusion and enhance drug/gene delivery, using both short (~μs) and long (~ms) ultrasound tone-bursts. Microbubble oscillation behavior can be observed with ultra-high speed microscopy; however, due to the high frame rate required and limited number of frames available, microbubble dynamic behavior can only be observed for a few acoustic cycles (~μs). In this report, the fate of microbubbles throughout long tone-burst ultrasound exposures (5 ms) was explored via direct microscopic optical observation in conjunction with passive cavitation detection, over a range of acoustic pressures (0.25, 0.5, 1.0, and 1.5 MPa) and microbubble concentrations (2e6, 2e7, and 2e8 MB/ml). The ultrasound system was triggered from UPMC Cam, the ultra-high speed camera system at the University of Pittsburgh Medical Center, such that microbubble activity at various time points (0–5 ms) within the long tone-burst was captured. Microbubbles first underwent stable or inertial cavitation depending on the acoustic pressure used, and then formed gas-filled clusters that continued to oscillate, break up, and form new clusters. Cavitation detection confirmed continued, albeit diminishing acoustic activity throughout the 5 ms ultrasound excitation. This discovery suggests that persisting cavitation activity during long tone-bursts may confer additional therapeutic effect for ultrasound-microbubble mediated therapies.

10:45

2aBA9. Radiation damping of an arbitrarily shaped bubble. Kyle S. Spratt, Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, 4307 Caswell Ave. APT E, Austin, TX 78751, sprattkyle@gmail.com), Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), Mark S. Wochner (AdBm Technologies, Austin, TX), and Mark F. Hamilton (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, California)

Large encapsulated bubbles have recently been used for abating low-frequency anthropogenic underwater noise [J. Acoust. Soc. Am. **135**, 1700–1708 (2014)]. The use of encapsulation allows for the possibility of bubbles that are significantly nonspherical in their equilibrium state. Strasberg [J. Acoust. Soc. Am. **25**, 536–537 (1953)] investigated the resonance frequency of an ideal bubble with arbitrary shape and found that the dependence of resonance frequency on the shape of the bubble reduced to a well-known problem in electrostatics. The present work extends that analysis to include the effects of radiation damping on the oscillation of a bubble, and does so by including a loss term due to Ilinskii and Zabolotskaya [J. Acoust. Soc. Am. **92**, 2837–2841 (1992)] in the volume-frame dynamical equation for the bubble. An expression is given for the amplitude of the acoustic field scattered from the bubble, and it is shown that radiation damping scales as resonance frequency cubed for arbitrarily shaped bubbles having the same volume. Comparisons are made with previous work on scattering from spherical and prolate spheroidal bubbles, and various new bubble shapes are considered. [Work supported by AdBm Technologies, the ARL:UT McKinney Fellowship in Acoustics and ONR.]

11:00

2aBA10. Effect of blood viscosity and thrombus composition on sonoreperfusion efficacy. John Black, Frederick Schnatz, Francois T. Yu, Xucai Chen, Judith Brands, Flordeliza Villanueva, and John Pacella (Heart and Vascular Inst., Univ. of Pittsburgh, 3550 Terrace St., Scaife Hall S960, Pittsburgh, PA 15261, yuft@upmc.edu)

Embolization during stenting for myocardial infarction (MI) causes microvascular obstruction (MVO). We demonstrated that ultrasound (US) and microbubble (MB) therapy [sonoreperfusion (SRP)] resolves MVO from venous microthrombi at plasma viscosity. However, blood is more viscous than plasma, and arterial microthrombi are mechanically distinct from venous clot. We tested whether blood viscosity and arterial microthrombi decrease SRP efficacy in our *in vitro* model of MVO. Lipid MB in plasma and blood viscosity PBS were passed through a 40 μm pore mesh. Arterial microthrombi were formed in a Chandler loop. Venous microthrombi were made statically. MVO was created by occluding the mesh with microthrombi until upstream pressure reached 40 mmHg. US (1 MHz; 5000 cycles; 0.33 Hz PRF; 0.6, 1.0, and 1.5 MPa peak negative pressure) was delivered with a single focused element transducer. MVO caused by arterial thrombi at increased viscosity resulted in less upstream pressure drop during therapy. PCD showed a decrease in inertial cavitation (IC) when viscosity was increased. SRP efficacy is less with arterial thrombi compared to venous thrombi, and higher viscosity further reduces SRP efficacy by decreasing IC. These findings could help guide the selection of US parameters for *in vivo* optimization of SRP.

11:15

2aBA11. Subharmonic threshold quantification of ultrasound contrast agents. Rintaro Hayashi, John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu), Parag Chitnis (BioEng., George Mason Univ., Fairfax, VA), and Jeffrey A. Ketterling (Riverside Res., New York, NY)

The measurement and prediction of the subharmonic threshold of ultrasound contrast agents is of significant interest, particularly for high frequency applications. Theoretical analytical predictions typically follow from a weakly nonlinear analysis assuming continuous, single frequency forcing. Furthermore, numerical simulation and experimental definitions are often based on the quantification of spectral response of the nonlinear harmonics. Limitations of these approaches are investigated with respect to pulsed forcing and associated non-stationary (chirp) excitations. Novel quantification and definitions are proposed with respect to instantaneous frequency and relative energy content of an empirical mode decomposition of the scattered pressure. Teager-Kasiser energy operator allows for additional quantification. The methodology is examined with respect to experimental polymer contrast agent data.

11:30

2aBA12. Stress and strain fields produced by violent bubble collapse. Lauren Mancia (Mech. Eng., Univ. of Michigan, 2016 Walter E. Lay Automotive Lab, Ann Arbor, MI 48109-2133, lamancha@umich.edu), Eli Vlaisavljevich (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), Matthew Warnez (Mech. Eng., Univ. of Michigan, Ann Arbor, MI), Zhen Xu (Biomedical Eng., Univ. of Michigan, Ann Arbor, MI), and Eric Johnsen (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Cavitation finds a key application in therapeutic ultrasound. For example, histotripsy relies on the rapid expansion of cavitation bubbles to fractionate soft tissue. To fully understand the mechanisms responsible for tissue fractionation, we numerically model cavitation in a tissuelike medium, focusing on the effect of its viscoelastic properties (viscosity, elasticity, and relaxation). It is hypothesized that ablation is caused by high strain rates and stresses exerted on the surrounding tissue as bubbles rapidly expand from nanometer to micron scales. The present study uses robust numerical techniques to compute the stress fields in the surrounding medium produced by single-bubble expansion. Bubble expansion is driven by a waveform that approximates a histotripsy pulse with relevant parameters, and soft tissue surrounding the bubble is modeled as a viscoelastic medium with Neo-Hookean elasticity. We will examine the stress, strain, and temperature fields produced during this process to explain potential damage mechanisms.

Session 2aNSa

Noise, Psychological and Physiological Acoustics and ASA Committee on Standards: Noise, Soundscape, and Health

Brigitte Schulte-Fortkamp, Cochair
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Klaus Genuit, Cochair
HEAD acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany

Chair's Introduction—8:30

Invited Papers

8:35

2aNSa1. Soundscape as a resource to balance the quality of an acoustic environment. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

Impacts of noise on people's quality of life, on their health, and on the quality of the living environment, including economic costs will be addressed in this paper. It follows the understanding of potential health effects related to noise by the European Environment Agency. "Adverse effects of noise occur when intended activities of the individual are disturbed. The sound level of the acoustic stimulus, its psychoacoustical sound characteristics, the time of its occurrence, its time course, its frequency spectrum, and its informational content will modify the reaction." EEA 2010. To account for the diversity of living situations related research will focus on the cooperation of human/social sciences like. psychology, sociology, architecture, anthropology, and medicine. Soundscape will provide the needed paradigm shift in research and evaluation. The International Standard ISO/DIS 12913-1 has as its purpose the enabling of a broad international consensus on the definition of "soundscape" and its respective evaluation. It is more than urgent to understand that there is the need to provide a solid foundation for communication across disciplines and professions with an interest in achievements on better solutions for the people concerned.

8:55

2aNSa2. The character of noise and its relation to noise effects. André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de)

The existence of noise effects leading to adverse health effects is beyond controversy. However, the main factors responsible for specific noise effects are still not clear. The human hearing system does not only register the sound pressure level of noise. It shows a high performance in amplitude, frequency, and time resolution and recognizes patterns in noise. Interestingly, several physiological reactions are not only influenced by the sound pressure level of noise. Due to certain psychoacoustic properties of noise or the level of noise annoyance stronger physiological reactions are stimulated. In order to predict responses to noise more reliably, psychoacoustic parameters and further hearing related-parameters have to be applied. Moreover, it is decisive to distinguish between auditory sensation and sound perception. Sound perception refers to complex processes, where additional information is processed and "mixed" with basic auditory sensations. Sound perception implies the cognitive processing, interpretation, and assessment of sound. Some studies suggest that the sound perception dimension is highly relevant to evoked physiological reactions and noise effects. The paper illustrates the difference between auditory sensation and sound perception and highlights the need to analyze sound in terms of perception to understand noise effects more in detail.

9:15

2aNSa3. The operating theater—A paying challenge for soundscape design. Holger Sauer (Inst. for Medical PsychoPhys., Klinikum Westfalen, Klinik am Park, Brechtener Straße 59, Lünen D-44536, Germany, holger.sauer@klinikum-westfalen.de)

The operating theater (OT) is virtually a parade example of a place with a high stress potential—not only for the working personnel but also for the patients. The reasons for this phenomenon are complex, and, partially inescapable. But since evidence reveals more and more the devastating consequences of this kind of stress for all persons concerned, there are good reasons to eliminate or at least reduce as many stress factors as possible. The soundscape is one of the major items which can work both stressing and de-stressing, however, quite diverging and dependant on the need and preferences of the several players. It is not sufficient simply to reduce the acoustic level since sound has different implications for the respective persons. Instead, the OT of the future requires a differentiated soundscape—implying hardware, software, and strategical patterns—which can be used in a more individualized way and, in addition, is adaptable to the respective circumstances. In order to achieve this, major efforts are required, among others due to complex preconditions; but it can be expected that these efforts will yield worthwhile results, which even will be transferable to other stress-sensitive domains.

9:35

2aNSa4. Soundscapes and human restoration in green urban areas. Irene van Kamp, Elise van Kempen, Wim Swart, and Hanneke Kruize (Sustainability, Environment and Health, National Inst. Public Health and the Environment, PO Box 1 Postbus 10, Bilthoven, Utrecht 3720 BA, Netherlands, irene.van.kamp@rivm.nl)

There is increasing interest in the association between landscapes, green and blue space, open countryside and human well-being, quality of life, and health. Most studies in this field do not account for the positive or negative moderating effects of the acoustic environment. This is partly due to the lack of relevant data, although basic models do refer to the role of traffic related noise and air-pollution. This paper reports on the results of a European study (Phenotype) into the health effect of access to and use of green area in four European cities. At the four study centers, people were selected from neighborhoods with varying levels of socioeconomic status and green space. By structured interview, information was gathered about availability, use, and importance of green space in the immediate environment, as well as the sound quality of favorite green areas used for physical activity, social encounters, and relaxation. Data are also available about perceived mental and physical health and medication use. This allows for analyzing the association between indicators of green and health, while accounting for perceived soundscapes. Audit data about the sound quality are also available at neighborhood level as a point of reference.

9:55

2aNSa5. Acoustic preservation: Preventing unseen loss in our historic sites. Pamela Jordan (Dept. of Eng. Acoust., Humboldt Fellow, Tech. Univ. of Berlin, Kremmener Strasse 15a, Berlin 10435, Germany, pam.f.jordan@gmail.com)

Soundscape research examines the vital role that sound plays within the integrated complexity of any specific environment, be it a rural landscape, cityscape, or indoor scenario. Historic preservation, on the other hand, generally seeks to safeguard shared built histories and frames the physical conservation of a site by prioritizing certain elements or activities for contemporary audiences. These designed interventions function similarly whether they are in "nature" or in the attic of a building. Yet the acoustic dimension of a site is seldom considered a vital part of preservation work. Even more rarely is the aural environment considered a contributing historic resource. Nevertheless, the soundscape of a place is an important cultural resource to learn from and protect—one that can potentially unlock a new dimension of understanding in a site's past. This submission presents current work being carried out in Berlin that bridges the gap between preservation practice and soundscape research. Acoustic concerns specific to historic sites will be discussed through case study examples. A discussion of developing methodologies for mapping soundscapes of historic sites will follow.

10:15–10:30 Break

Contributed Papers

10:30

2aNSa6. Soundlapse: Audio strategies for documenting soundscapes with timelapse photography. J. Parkman Carter (Architectural Acoust., Rensselaer Polytechnic Inst., 32204 Waters View Circle, Cohoes, NY 12047, cartej8@rpi.edu)

Timelapse photography—the procedure of taking photographs at regular intervals and then sequencing them together at a rate typical of film/video (e.g., 24 frames per second) to produce a radically accelerated and compressed portrayal of time passing—is widely popular across such diverse areas as scientific research, weather tracking, filmmaking, and even reality TV shows, which commonly use timelapse footage to segue between scenes and reflect the passing of time. The compelling visual effect and broad legibility of timelapse photography makes it a powerful tool for revealing flows, shifts, and patterns in the environment, which would otherwise be too subtle to perceive in real time. One of the most important challenges in documenting the salient features of a soundscape is revealing similar shifts and patterns across time in the acoustic environment. However, temporal logic would suggest it is simply not possible to sequence such brief samples of the audio environment with any meaningful result. This project explores different audio capture and editing techniques to produce viable strategies for documenting the changes in a soundscape which accompany timelapse photography of the environment under study. Examples will be demonstrated, emphasizing the significant variables for adequately representing the elusive temporal characteristics of a soundscape.

10:45

2aNSa7. The effect of buffering noise in passenger car. Miguel Iglesias, Ismael Lengua, and Larisa Dunai (Dept. of Graphic Eng., Universitat Politècnica de València, St. Camino de Vera s/n 5L, Valencia 46022, Spain, inecav@inecav.com)

This paper presents the effect of buffering noise in passenger cars in real conditions at variable speed. At 25% of opened back window, the effect of buffering noise is not perceived. When increasing the aperture of the window to 50%, is perceived the creation of the buffering noise effect with resonance frequency = 15.6 Hz at speed = 80 km/h, but is not irritating because the pressure level is low, pressure level = 110.2 dB. At 75% of the window aperture appears two air flows in the inside car: the one coming from the superior part of the superior zone of the inside and another one from the central zone of the inside. The effect of buffering noise is clearly perceived with resonance frequency = 16.4 kHz at speed = 80 km/h and pressure level = 119 dB and 19.5 Hz at speed = 110 km/h and pressure level at 122 dB. At speed higher than 103–140 km/h, the buffering noise disappears, depending on the car model. For Opel Corsa, the effect of buffering noises appears at the same speed range, whereas for Peugeot and BMW models, it appears three ranges, one from the buffering noise creation, persist in another range, and disappear in the third range.

11:00

2aNSa8. Innovative tools for urban soundscape quality: Real-time road traffic auralization and low height noise barriers. Alexandre Jolibois, Jérôme Defrance, Julien Maillard, Philippe Jean, and Jan Jagla (CSTB (Ctr. Scientifique et Technique du Bâtiment), CSTB, 24 rue Joseph Fourier, Saint-Martin-d'Hères 38400, France, alexandre.jolibois@cstb.fr)

Although noise exposure has been unanimously recognized for its impacts on people's health, noise is still a current problem in many cities across the world. Besides, most noise analysis tools are based on long term average levels, which were initially meant to be used for inter-city infrastructures and are therefore insufficient to quantify the soundscape quality in urban environments. In this paper, we present some of the recent developments and results regarding two classes of innovative tools dedicated to the improvement of urban soundscape quality. First, recent advances in the field of real-time auralization of road traffic noise are presented and the benefit of realistic sound field restitution as a new analysis tool for the evaluation of urban planning projects is discussed and demonstrated. Second, some promising results regarding innovative urban noise reducing devices, taken among other works from the European project HOSANNA, are presented. In particular, the potential of vegetation and low height noise barriers are emphasized, from a numerical and experimental point of view. Results show that these devices can significantly enhance the soundscape quality in specific places, and therefore can help in the creation of so-called "quiet areas" within urban environments.

11:15–11:35 Panel Discussion

Session 2aNSb

Noise and Structural Acoustics and Vibration: Active Control of Sound and Vibration

Scott D. Sommerfeldt, Cochair
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Pegah Aslani, Cochair
Physics and Astronomy, Brigham Young University, N203 ESC, Provo, UT 84602-4673

Chair's Introduction—9:00

Invited Papers

9:05

2aNSb1. A feedback active noise control for in-ear headphone with robust on-line secondary path modeling. Young-cheol Park, Keun-sang Lee, and Youna Ji (Comput. & Telecomm Eng Div., Yonsei Univ., Changjo-hall 269, Yonseidae-ro 1, Wonju, Gangwon-do 220-710, South Korea, young00@yonsei.ac.kr)

The active noise control (ANC) technique can be used to improve the comfortness and quality of the music listening with an in-ear headphone. For a compact headphone design, the feedback ANC operated with the filtered-x LMS (FxLMS) algorithm is preferred. In practice, the error in the estimated secondary path can be problematic for the FxLMS algorithm, since it can lead the algorithm to an unstable state, and thus cause degradation of sound quality with the in-ear headphone. In this study, an adaptive residual music canceler (RMC) is proposed for enhancing the accuracy of the reference signal of the feedback ANC. Since RMC is designed to track the bias of the current secondary path estimate, the secondary path is continuously updated by combining the previous estimate of the secondary path with the current weight vector of RMC. In addition, variable step-size schemes are developed for both the control and secondary path estimation filters, which enable the ANC system to adapt quickly and robustly to the variation of the physical secondary path. Simulation results show that the secondary path can be accurately estimated and high quality of music sound can be consistently obtained in a time-varying secondary path situation.

9:25

2aNSb2. Real-time implementation of multi-channel active noise control systems for infant incubators. Lichuan Liu and Xianwen Wu (Electrical Eng., Northern Illinois Univ., 590 Garden Rd., Dekalb, IL 60510, liu@niu.edu)

High noise in infant incubators results in numerous adverse health effects, which have lifelong consequences for neonatal intensive care unit (NICU) graduates. This paper presents multi-channel feed forward active noise control (ANC) systems for infant incubators. Then, the proposed algorithm is implemented by using Texas Instrument TMS320 C6713. The real-time experiment results show that the proposed algorithm is effective in canceling the noises of infant incubators in real NICU environment

9:45

2aNSb3. Sweeping tunable vibration absorbers for structural vibration control. Paolo Gardonio and Michele Zilletti (DIEGM, Univ. of Udine, Via delle Scienze, 208, Udine 33100, Italy, paolo.gardonio@uniud.it)

Tuneable vibration absorbers (TVAs) have been successfully implemented in distributed thin structures (e.g., panels shells) to control either tonal vibration, produced, for example, by unbalanced machinery, or low-frequencies broadband vibration, generated, for example, by stochastic pressure distributions due to diffuse sound fields or turbulent boundary layer fluid flows. This study is focused on the broadband control of structural vibration where often multiple TVAs are used to control the response of the natural modes that resonate in the controlled frequency band. Normally, each TVA is tuned to minimize the resonant response of a specific natural mode of the hosting structure by tuning the TVA natural frequency to the resonance frequency of the mode and setting the TVA damping ratio to properly dissipate energy. The proposed sweeping tuneable vibration absorbers (STVAs) have been conceived to operate on the multiple natural modes of the hosting structure where they are mounted by sweeping the TVA tuning frequency over the frequency band of control and simultaneously varying the TVA damping ratio to effectively dissipate energy over the desired frequency range. The operation principle of a single STVA is first discussed and demonstrated experimentally. The implementation of multiple STVAs is then examined.

10:05–10:25 Break

10:25

2aNSb4. Active control of cylindrical shells using the weighted sum of spatial gradients control metric. Pegah Aslani, Scott D. Sommerfeldt (Phys. and Astronomy, Brigham Young Univ., N203 ESC, Provo, UT 84602-4673, pegah.aslani@gmail.com), and Jonathan D. Blotter (Mech. Eng., Brigham Young Univ., Provo, UT)

Often it is desired to reduce the sound radiated from vibrating structures, including cylindrical shells. Active structural acoustic control (ASAC) provides a means to control the structural radiation at low frequencies efficiently and effectively. The technique of using the weighted sum of spatial gradients (WSSG) as a control metric has been developed previously for flat structures. This paper will investigate control of WSSG for cylindrical shells. There are specific features associated with WSSG that tend to provide effective control of radiated sound power. The method has also been shown to be quite robust with respect to error sensor location. The effectiveness of WSSG control has been investigated through the use of radiation modes for cylindrical shells, which allows us to determine the radiated sound power both before and after control. Results using WSSG control will be reported, along with comparisons of the results obtained with some of the other possible control approaches reported previously.

10:45

2aNSb5. Adaptive vibration control of a mechanical system with nonlinear damping excited by a tonal disturbance. Michele Zilletti (Inst. of Sound and Vib. Res., Univ. of Southampton, Via delle Scienze, 206, Udine 33100, Italy, michele.zilletti@uniud.it), Stephen J. Elliott, and Maryam Ghandchi Tehrani (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom)

In this study, an adaptive control system to reduce the tonal vibration of a mechanical system characterized by nonlinear damping is considered. Since the response of the system to a sinusoidal excitation is however mainly sinusoidal, nonlinear controllers are not required for the control of a tone in such a mechanical system. The adaptation and stability of an adaptive control algorithm, however, depends on the accuracy of the plant model. Since the tonal response of the nonlinear system changes with excitation level, conventional adaptive algorithms, with a fix linear model of the plant, can be slow to converge and may not achieve the desired performance. The use of an online observer is proposed, to estimate the describing function model of the plant, which will vary with excitation level. This allows the adaptive control algorithm to converge more quickly than using a fixed plant model, although care has to be taken to ensure that the dynamics of the observer do not interfere with the dynamics of the adaptive controller.

Contributed Papers

11:05

2aNSb6. A study on analysis of room equalizer using smart modules for analog high-quality audio. Bae Seonggeon (Soongsil Univ., 1118-408, Mokdong Apt. 11 Dangi, Yangcheon Gu, Seoul 148-771, South Korea, sgbae123@gmail.com), Kim Jaepyung (Daelim Univ., Seoul, South Korea), and Rhee Esther (Keimyung Univ., DaeGu, South Korea)

This study proposes room equalizer, which uses movable and variable modules and revises frequency response, and is also applicable to a variety of places. When equalizing frequency responses, commonly used electric graphic equalizers or parametric equalizers have the same effect as the sound signals such as audio passing through electrical elements like condenser or coil, and phase shift of audio signal occurs at this time. Accordingly, they secure frequency response by adjusting and equalizing the characteristics of sound-absorbing materials without any distortion of the original sound. Therefore, this study proposes a room equalizer that does not use these electrical characteristics, but uses non-electronic spatial matching with an excellent sound persistency rate to enable its application to various changes in space.

11:20

2aNSb7. Thermoacoustic approach for the cloaking of a vibrating surface. Avshalom Manela and Leonid Pogorelyuk (Aerosp. Eng., Technion, Technion City, Haifa 32000, Israel, amanela@technion.ac.il)

A vibrating surface in a quiescent fluid transmits pressure fluctuations, which propagate into far-field sound. The vibroacoustic mechanism involved, coupling vibration and sound, is common in a large number of applications. Yet, there is a growing interest in developing means for achieving "acoustic cloaking" of an animated boundary. We suggest the heating of a surface, generating thermoacoustic perturbations, as a mechanism for monitoring vibroacoustic sound. Considering a setup of an infinite planar wall interacting with a semi-infinite expanse of an ideal gas, we investigate the system response to arbitrary (small-amplitude) vibro-thermal excitation of the confining wall. Analysis is based on continuum Navier-Stokes-Fourier and kinetic Boltzmann equations, and supported by stochastic direct simulation Monte Carlo calculations. Starting with a case of a sinusoidally excited boundary, a closed-form solution is derived in both continuum and collision-free limits. The results, found valid at a wide range of frequencies, indicate that effective cloaking of the boundary may be achieved through optimal choice of boundary-generated heat flux. The results are rationalized by considering the quasistatic limit of low-frequency excitation. The analysis is then extended to consider the system response to impulse (delta-function) actuation, where the independence of cloaking conditions on perturbation frequency is demonstrated.

11:35

2aNSb8. Monitoring floor impact sounds using detection algorithms in multi-story residential buildings. Jin Yong Jeon, Joo Young Hong, Hansol Lim, and Muhammad Imran (Architectural Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, jyjeon@hanyang.ac.kr)

A monitoring and alarming system was designed to detect household noise occurrence in the multi-story residential buildings. This system is composed of a microphone combined with a home smart pad installed on a wall and a main server that can store data to discriminate household noise occurrence. Sound levels such as L_{Aeq} and L_{Amax} were calculated in real time using developed digital signal processor (DSP) of the acoustical parameter analyzer in the home smart pad. A correction equation was applied based on the standard measurement method to provide accurate measurements. After measurement, the results were saved in the main server using Ethernet and inter-compared with the data of each household to discriminate the household noise occurrence. Finally, the alarm was raised depending on the decision criteria based on both the regulations of Korean government and subjective evaluation results from the previous study.

Session 2aPA

Physical Acoustics: Wind Noise Mitigation in Atmospheric Acoustics

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JohnPaul Abbott, Cochair
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Chair's Introduction—8:00

Invited Papers

8:05

2aPA1. The prediction of infrasonic wind noise in forests. Richard Raspet and Jeremy Webster (NCPA, Univ. of MS, PO Box 1848, NCPA, University, MS 38677, raspet@olemiss.edu)

Recent research on the measurement and prediction of the infrasonic wind noise measured in pine and mixed deciduous forests with and without leaves has led to a good understanding of the turbulence mechanisms that generate the infrasonic wind noise. This talk will review the success of the predictions of the wind noise from the measured horizontal velocity spectrum above the canopy and in the canopy as well as the wind velocity profile above and in the canopy. The wind profiles measured in this study were not typical of those published in the meteorology literature due to their limited height and sparse regular planting. The height dependence of the 3-D turbulence spectra also differ from the model spectra. The modifications necessary to produce predictions for more typical forests from the meteorological literature will be described. [Work supported by the U. S. Army Research Office under grant W911NF-12-0547.]

8:25

2aPA2. Experimental quantification of microphone windscreens and noise cones in a low speed wind tunnel. Andrew R. Barnard (Mech. Eng. - Eng. Mech., Michigan Technolog. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, arbarnar@mtu.edu)

Wind noise is a common challenge in all types of atmospheric acoustics as well as in wind tunnel applications in the aerospace and automotive industry. The dynamic pressure field measured by the microphone includes both the acoustic pressure and the pressure induced by the turbulent air flow over the diaphragm. There are three types of microphone accessories that are commonly used to perform wind noise isolation: protective grid-caps (no isolation), windscreens, and nose-cones. In this study, each of these microphone accessories is tested in a wind tunnel using 1/4" and 1/2" microphones in turbulent and laminar flows at flow speeds up to 55 mph and in both the head-on and parallel diaphragm orientations. Two types of flush mount microphones, a surface mount and a side-vented pressure microphone, are also evaluated. The effects of background flow noise on the microphone measurement is shown for all conditions and a case study is presented with an arbitrary acoustic source. Finally, recommendations are presented for which accessories are best used in different measurement situations.

8:45

2aPA3. Wind noise suppression using compact arrays of infrasound sensors. William G. Frazier (Ste. 142, 850 Insight Park Ave., University, MS 38677, gfrazier@hyperiontg.com)

This presentation describes how a compact array (aperture much less than the shortest wavelengths of interest) of infrasound sensors can be used to significantly enhance signal detection and waveform estimation in the presence of high levels of wind noise without the use of unwieldy mechanical screening. The methodology's effectiveness is founded on the fact that wind noise can be highly correlated on short spatiotemporal scales. This correlation structure is adaptively estimated from data and is used to formulate a generalized likelihood ratio detection problem and a minimum mean squared error waveform estimation problem. The infrasoudnic waveform is explicitly represented by a user-definable, parametrically characterized, stochastic prior distribution. Choice of this prior can enhance detection of anticipated signals and suppress others and thus more than one prior can be utilized in order to perform some level of classification. The presentation provides typical performance results from a range of application scenarios. Application to more traditional infrasound arrays is also presented, illustrating that exploitation of only the temporal correlation properties of wind noise is also beneficial in the context of the stochastic models and estimation techniques utilized.

9:05

2aPA4. Pipe systems for infrasound wind-noise reduction: Issues and approaches. Thomas B. Gabrielson (Penn State Univ., PO Box 30, State College, PA 16804, tbg3@psu.edu)

Wind-noise-reduction for infrasound monitoring below 10 Hz often takes the form of multiple-inlet pipe networks that average the field over distances small with respect to the acoustic wavelength but large with respect to the turbulence scale. There is a long tradition of pipe systems; however, traditional approaches can result in mediocre performance. Three aspects of the traditional design—inlets arranged in a regular geometric pattern, resistive resonance suppression, and equal-length inlet-to-sensor paths—can all be questioned. This paper reviews pipe-system performance with respect to design, modeling, and data quality. Waveguide models that account for the acoustic adiabatic-to-isothermal transition inside the pipes and summing junctions have been validated for prediction of complex frequency response and horizontal and vertical directionality. Techniques have also been developed for *in-situ* measurement of system frequency response and wind-noise-reduction effectiveness. The importance of on-site measurement is underscored by the number of performance problems uncovered at operational infrasound monitoring sites by in-situ techniques. These issues will be illustrated with measurements at operational infrasound-monitoring stations and at test-bed systems.

9:25

2aPA5. Long duration wind noise abatement measurements at the University of Mississippi Field Station. Carrick L. Talmadge (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38655, clt@olemiss.edu)

A series of wind noise abatement tests were performed at the University of Mississippi Field Station from January through June of 2014. These tests compared bare sensors to sensors treated with porous hoses, and domes of different dimensions (40, 48, 60, and 96 in.ch) that were covered either with foam or with metal meshing (30% and 45% opening). Because these measurements were obtained continuously, measurements were obtained over a wide range of weather conditions. As expected, the frequency of maximum wind noise attenuation decreased with increasing diameter of the dome, and the maximum attenuation obtained increased commensurately with diameter. Tests were also performed with nested domes (96 and 40 in.ch), which provided greater attenuation than just the larder of the two domes. The 30% opening mesh proved superior to the 45% opening mesh. Generally, the porous hoses provided superior wind noise protection below 1-Hz, but did not provide as much wind noise attenuation as was provided by the 96 and 40 in.ch nested wind dome configuration. We found that a number of issues (e.g., less robust, more affected by wet conditions) make this a less idea solution than the metal mesh covered nested domes, for permanent installations.

Contributed Papers

9:45

2aPA6. Wind-induced noise based upon stability dependent turbulent velocity spectrum models. Carl R. Hart and D. Keith Wilson (U.S. Army Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, carl.r.hart@usace.army.mil)

The ambient sound level is an important datum for noise control, quantifying soundscapes, and remote detection. For low acoustic frequencies and fast enough wind speeds, wind-induced noise contributes largely to the measured ambient sound spectrum. Predictive models for wind-induced noise are based upon an empirical relationship between fluctuating pressure, mean wind speed, and the fluctuating streamwise wind velocity. A wind noise model developed by Van den Berg [Van den Berg, G. P., J. Acoust. Soc. Am. **119**(2), 824–833, 2006] is derived from a turbulent velocity spectrum, which is applicable for neutral atmospheric stability and flat homogeneous terrain. In this work, several other wind noise models are formulated based upon different turbulent velocity spectrum models. The turbulent velocity spectrum models take into account atmospheric stability, with varying degrees of complexity. Differences in predicted wind noise levels among the models show nearly a 3 dB spread for some atmospheric conditions. A recommendation is given for a specific wind noise model based on intercomparisons.

10:00–10:15 Break

10:15

2aPA7. Statistical properties of the mirror flow model of boundary-layer turbulence. Gregory W. Lyons and Nathan E. Murray (National Ctr. for Physical Acoust., The Univ. of MS, NCPA, P.O. Box 1848, University, MS 38677-1848, gwlyons@go.olemiss.edu)

The mirror flow model is a construction of inhomogeneous, anisotropic turbulence from the superposition of a homogeneous, isotropic turbulent field with itself in transformed coordinates. This model has been used for atmospheric turbulence to predict spectral contributions to low-frequency wind noise [Yu *et al.*, J. Acoust. Soc. Am. **129**(2), 622–632 (2011)]. Here, analytical solutions are reported for the longitudinal and vertical turbulence spectra as a function of elevation. The correlation functions, integral turbulence scale, and profiles of mean-square velocity components are computed. The premultiplied spectra show remarkable agreement with observed scaling in the atmospheric boundary layer. Implications for computation of wind noise spectra are discussed.

10:30

2aPA8. Effects of ground characteristics on wind-seismic coupling. Vahid Naderyan, Craig Hickey, and Richard Raspet (National Ctr. for Physical Acoust. and Dept. of Phys. and Astronomy, Univ. of MS, NCPA, 1 Coliseum Dr., University, MS 38677, vnaderya@go.olemiss.edu)

In seismic surveys, wind noise at low frequency seriously degrades seismic data quality. In order to find a solution for this problem, the driving pressure perturbations on the ground surface associated with wind-induced ground motions were investigated in a previous work [Naderyan, Hickey, and Raspet, J. Acoust. Soc. Am. **136**, 2139 (2014)]. Multiple triaxial 10 Hz geophones were deployed at different depths to study the induced ground velocity as a function of depth. The displacement amplitudes of the ground motions as a function of depth and frequency were predicted, where the ground was modeled as a homogeneous half-space elastic media. In the current work, the predictions are extended for inhomogeneous ground. The measurements are conducted at a different site with different characteristics. Since wind noise is dominant at lower frequencies, the triaxial 4.5 Hz geophones were used to investigate a lower frequency range. [This work was supported by USDA under award 58-6408-1-608.]

10:45

2aPA9. Comparisons of semi-porous fabric domes and cylindrical wind fence enclosures for infrasonic wind noise reduction. JohnPaul Abbott and Richard Raspet (NCPA and Dept of Phys. and Astronomy, Univ. of MS, 122 PR 3049, Oxford, MS 38655, jrabbott@go.olemiss.edu)

This investigation compares the measured wind noise of two types of infrasonic wind noise reduction barriers. The first type is a set of 2.0 m diameter semi-porous fabric domes constructed and tested by the Army Research Laboratory [J. Acoust. Soc. Am. **136**, 2139 (2014)]. The type second is a set of single and multi-layer cylindrically shaped variable-porous

wind fence enclosures measuring 5.0 m and 10.0 m, constructed and tested by the National Center for Physical Acoustics [J. Acoust. Soc. Am. **134**, 4161 (2013)]. Preliminary investigations showed that for wind speeds between 3—6 m/s the measured power spectra densities of the fabric domes were comparable to the measured power spectra densities of the single layer, 5.0 m diameter wind fence enclosures at mid-range porosities. The measured power spectra densities of the fabric domes were higher than the measured power spectra densities for the 10.0 m diameter and multi-layer wind fences enclosures. For wind speeds greater than 6 m/s, the measured wind noise for the fabric domes was comparable to or lower than the measured wind noise for the wind fence enclosures.

11:00

2aPA10. A comparison of sintered materials for the reduction of wind noise in acoustic sensors. Latasha Solomon, David Gonski, Stephen Tenney, and Leng Sim (US Army Res. Lab, 2800 Powder Mill RD, Adelphi, MD 20783, latasha.i.solomon.civ@mail.mil)

Wind noise reduction has been of particular interest in the acoustic sensor arena for many years. Ideally, a desirable noise-mitigating medium is resilient to extreme temperatures and dust, as well as insects/animals. Open cell polyurethane foam is the most widely chosen material for acoustic wind noise reduction. These windscreens tend to provide protection from humidity and moisture, as well as reduction of wind flow noise at the frequencies of interest (500 Hz and below). However, after prolonged use in the environment, degradation of the windscreen and its performance is inevitable. Exposure to high levels of UV rays tends to make the windscreens brittle, while sand and dirt may become lodged in the pores requiring frequent cleaning or replacement, and they can be vulnerable to wildlife. Ideally, one would like to replace the current foam windscreen with a more durable, smaller, lower-profile medium providing similar wind noise rejection to that of foam windscreen. This research will compare the effectiveness of porous metal and plastic windscreens fabricated from sintered material to that of the conventional open cell foam windscreen in reducing wind generated flow noise.

11:15

2aPA11. New calculation of the turbulence-turbulence contribution to wind noise pressure spectra by incorporating turbulence anisotropy. Jiao Yu, Yanying Zhu (Liaoning Shihua Univ., 1 Dandong Rd. West Section, Wanghua District, Fushun, Liaoning 113001, China, yujiaojoy@hotmail.com), and Richard Raspet (National Ctr. for Physical Acoust., University, MS)

Turbulence-turbulence interaction and turbulence-shear interaction are the sources of intrinsic pressure fluctuation for wind noise generated by atmospheric turbulence. In previous research [Yu *et al*., J. Acoust. Soc. Am. **129**(2), 622–632 (2011)], it was shown that the measured turbulent fields outdoors can be realistically modeled with Kraichnan's mirror flow model and turbulence-shear interaction pressure spectra at the surface were predicted and compared to measurements. This paper continues to apply Kraichnan's model and calculates the turbulence-turbulence interaction wind noise pressure spectra under anisotropic turbulence conditions. Different from turbulence-shear interaction, the fourth-order moments of the turbulence are needed in the calculation. A new calculation of the turbulence-turbulence contribution to the wind noise pressure spectra is compared to that for isotropic turbulence and the results are analyzed. [Work supported by the U. S. Army Research Office (Grant No. W911NF-12-0547) and the National Natural Science Foundation of China (Grant No. 11304137).]

11:30

2aPA12. Reduced-size windscreens for infrasound. W. C. Kirkpatrick Alberts, William D. Ludwig (US Army Reserach Lab., 2800 Powder Mill, Adelphi, MD 20783, kirkalberts@verizon.net), and Carrick Talmadge (National Ctr. for Physical Acoust., University, MS)

The long wavelengths of infrasound necessitate large, often complex, wind noise reduction apparatus, e.g., porous hose, pipe arrays, or wind fences. For the infrasonic sources of military interest that fall between approximately 2 and 20 Hz, 6-m radius porous hose rosettes are often sufficient windscreens. However, the number of hoses required for a typical 4-element infrasound array (16–24) reduces its portability. Based upon recent research demonstrating that porous fabric domes could outperform porous hoses in the 2–20 Hz region and a need for more compact windscreens, 1.2-m diameter domes covered in three types of porous fabric and one folding perforated aluminum dome were investigated for their suitability as alternatives to porous hose. Evaluations of wind noise reduction, transfer functions, and environmental durability will be presented for these alternative windscreens.

Session 2aPP

Psychological and Physiological Acoustics, Biomedical Acoustics, Speech Communication, and Signal Processing in Acoustics: Celebration of the Modern Cochlear Implant and the First Substantial Restoration of a Human Sense Using a Medical Intervention I

Michael Dorman, Cochair
Department of Speech and Hearing Science, Arizona State University, Tempe, AZ 85258

Blake S. Wilson, Cochair
Duke University, 2410 Wrightwood Ave., Durham, NC 27705

Chair's Introduction—8:30

Invited Papers

8:35

2aPP1. From W. House to the present: A brief history of cochlear implants. Michael Dorman (Dept. of Speech and Hearing Sci., Arizona State Univ., Tempe, AZ 85258, mdorman@asu.edu)

The history of cochlear implants can be divided into four stages. The first, in the early 1960s, was the period of the pioneers (e.g., House, Doyle, and Simmons) who found minimal interest, even distain, from others for their work. The second stage, in the 1970s, saw a worldwide proliferation of groups who designed devices with multiple sites of cochlear stimulation and minimally effective, but very useful, speech coding strategies. The third period (1983–2003) saw the development of more effective processing strategies that utilized multiple sites of stimulation. These strategies, or their derivatives, are the ones in use today. Unfortunately, there have been no gains in group mean word understanding scores for patients fit with a single cochlear implant over a period of about 20 years. For that reason, in the current, fourth stage, researchers have improved performance by adding information (i) from residual low-frequency hearing (under 500 Hz) either in the implanted ear or in the ear contralateral to the implant or (ii) from a second implant. In this talk, I will view these stages from the standpoint of speech acoustics and the seemingly minimal amount of information that is sufficient to support high levels of speech understanding.

8:55

2aPP2. The punctuation mark in an equilibrium state: Modern signal processing. Blake S. Wilson (Duke Univ., 2410 Wrightwood Ave., Durham, NC 27705, blake.wilson@duke.edu)

As recently as the late 1980s, the speech reception performance of cochlear implants (CIs) was relatively poor compared to today's performance. As noted in the 1988 NIH Consensus Statement on CIs, only about 1 in 20 users of the best CIs at the time could carry out a normal conversation without the aid of lipreading. In contrast, the great majority of today's users can understand speech in relatively quiet conditions with their CIs and restored hearing alone. Indeed, most users communicate routinely via telephone conversations. As noted in the 1995 NIH Consensus Statement on CIs in Adults and Children, "A majority of those individuals with the latest speech processors for their implants will score above 80 percent correct on high context sentences, even without visual cues." Such abilities are a long trip from total or nearly total deafness. In this talk, I will (1) present some historical aspects; (2) describe the jump up in performance that was achieved in 1989 and provided the basis for many subsequent developments; and (3) mention some possibilities for further improvements in these already marvelous devices.

9:15

2aPP3. Additions to a single CI to improve speech understanding. René H. Gifford (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, 9302 MCE South, Nashville, TN 37232, rene.gifford@Vanderbilt.edu), Louise Loiselle (MED-EL, Durham, NC), Sarah Cook (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ), Timothy J. Davis, Sterling W. Sheffield (Hearing and Speech Sci., Vanderbilt Univ., Nashville, TN), and Michael F. Dorman (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ)

Speech understanding was assessed in a simulated restaurant and cocktail party environment for 39 adult cochlear implant recipients (18 bilateral, 21 hearing preservation). Speech was presented as either collocated (S_0N_0) or spatially separated from the noise ($S_0N_{0\text{-}360}$ or $S_0N_{90\&270}$). We calculated the benefit of adding a second CI, acoustic hearing from the non-implanted ear, and the implanted ear as well as spatial release from masking. The addition of either a second CI or binaural acoustic hearing yielded mean benefit of 13-percentage points. The additional benefit afforded via acoustic hearing in the implanted ear in addition to the bimodal configuration was an additional 10-percentage points. Little-to-no spatial release from masking was noted for either the simulated restaurant or the cocktail party environments. Spatial release from masking was not found to be significant at the group level nor was it different across the hearing configurations. We draw two primary conclusions: (1) adding either a second CI or binaural acoustic hearing to a single CI yields similar levels of benefit, but with different underlying mechanisms, and (2) the lack of spatial release is likely related to the presence of noise at both ears directed toward the ports of the HA and CI microphones.

9:35

2aPP4. Inserting the speech signal above the cochlea: How can this work? Colette M. McKay (Bionics Inst., 384-388 Albert St., East Melbourne, Victoria 3002, Australia, cmckay@bionicsinstitute.org)

Although cochlear implants (CIs) have successfully restoring hearing to many deaf people, there remains a subgroup who cannot benefit from them due to malformation of the cochlea or damage to the cochlear nerve. Auditory brainstem implants (ABIs) and auditory midbrain implants (AMIs) aim to restore hearing to this group. ABIs stimulate the surface of the cochlear nucleus (CN). Despite some ABI users gaining benefit similar to a CI, the majority achieve poor speech understanding with the device alone. The reasons for poor outcomes are not fully understood but include tumor damage to particular cells in the cochlear nucleus, surgical approach, and placement of the electrodes. One factor in poor outcome is the difficulty of stimulating specific tonotopic planes within the CN. This factor and tumor damage to the CN were reasons for the development of the AMI, which directly stimulates the tonotopic layers of the inferior colliculus (IC) using a shank electrode. Psychophysics and physiological studies have thrown light on the way that new electrode designs and processing strategies may provide patients with AMIs and potentially ABIs with hearing commensurate with CIs. [The Bionics Institute acknowledges the support it receives from the Victorian Government through its Operational Infrastructure Support Program.]

9:55

2aPP5. Neuroplasticity in deafness: Evidence from studies of patients with cochlear implants. Anu Sharma (Speech Lang. and Hearing Sci., Univ. of Colorado at Boulder, 2501 Kittredge Loop Rd., Boulder, CO 80309, anu.sharma@colorado.edu)

Deafness alters the normal connectivity needed for an optimally functioning sensory system—resulting in deficits in speech perception and cognitive functioning. Cochlear implants have been a highly successful intervention because they bypass cochlear damage and directly stimulate the auditory nerve and brain, taking advantage of high degree of neuroplasticity of the auditory cortex. Deaf adults and children who receive intervention with cochlear implants have provided a unique platform to examine the trajectories and characteristics of deprivation-induced and experience-dependent neuroplasticity in the central auditory system. I will describe changes in neural resource allocation secondary to hearing impairment, cross-modal cortical re-organization from the visual and somatosensory modalities, and the multimodal and cognitive re-organization that results from auditory deprivation. Overall, it appears that the functional activation of cognitive circuitry resulting from cortical reorganization in deafness is predictive of outcomes after intervention with electrical stimulation. A better understanding of cortical functioning and reorganization in auditory deprivation has important clinical implications for optimal intervention and re-habilitation of cochlear implanted patients. [Work supported by NIH.]

10:15

2aPP6. Language emergence in early-implanted children. Ann E. Geers (Behavioral and Brain Sci., Univ. of Texas at Dallas, 800 W Campbell Rd., Richardson, TX 75080-3021, ageers@utdallas.edu), Johanna G. Nichols (Otolaryngol., Washington Univ., St. Louis, MO), Emily A. Tobey (Behavioral and Brain Sci., Univ. of Texas at Dallas, Dallas, TX), and Lisa Davidson (Otolaryngol., Washington Univ., St. Louis, MO)

This study develops a model to differentiate early-implanted children with preschool language delays that persist into the elementary grades from children who achieve normal language as they gain auditory, linguistic, and academic experience. A nationwide sample of 60 children with congenital profound deafness, auditory-oral education, and early cochlear implantation (12–36 months) was assessed at average ages of 3.5, 4.5, and 10.5 years. Children were classified with: Normal Language Emergence (NLE: $N = 19$), Late Language Emergence (LLE: $N = 22$), or Persistent Language Delay (PLD: $N = 19$) based on standardized language test scores. Children with PLD did not differ from children with LLE on a number of variables that have previously been associated with post-implant outcome, including pre-implant hearing, age at first CI, gender, maternal education and nonverbal intelligence. Logistic regression analysis identified variables that predicted whether children with early language delay would catch up with hearing age-mates: (1) preschool speech and language level, (2) ear placement of first CI, (3) upgrades in speech processor technology, and (4) CI aided thresholds. A majority of the 60 children in this study reached age-appropriate levels of language, reading, and verbal reasoning and were placed in primary grades with hearing age-mates.

10:35–10:50 Break

10:50

2aPP7. Pitch perception and representations of frequency in the peripheral auditory system: What's missing in cochlear implants? Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Pitch is a crucial feature of auditory perception. It conveys information about melody and harmony in music, and conveys prosodic and (in tone languages) lexical information in speech. Although pitch has been studied formally for many decades, there still remains considerable uncertainty regarding how frequency and pitch information is coded and represented in the auditory periphery. This uncertainty means it is unclear whether and how pitch information can best be conveyed via cochlear implants. Cochlear-implant users have been shown to be sensitive to changes in the place of stimulation within the cochlea ("place" pitch) and to changes in the pulse rate presented to the electrode ("rate" pitch), at least up to rates of around 300 Hz. Place pitch in cochlear implants is probably best compared to "brightness" and rate pitch has most in common with the relatively weak pitch produced by temporal-envelope fluctuations in normal-hearing listeners. Neither type of pitch seems to provide the salient and accurate pitch sensations associated with normal hearing. Studies in both normal-hearing listeners and cochlear-implant users will be reviewed to shed light on the mechanisms of pitch perception and on the challenges involved in restoring pitch to cochlear-implant users. [Work supported by NIH grant R01DC005216.]

11:10

2aPP8. The sound of a cochlear implant: Lessons learned from unilaterally deaf patients fit with a cochlear implant. Michael Dorman, Sarah Cook (Dept. of Speech and Hearing Sci., Arizona State University, Tempe, AZ 85258, mdorman@asu.edu), and Daniel Zeitler (Denver Ear Assoc., Englewood, CO)

Starting about 10 years ago in Europe, patients who had one normally hearing ear and one deafened ear were fitted with cochlear implants in an effort to suppress tinnitus in the deafened ear. That effort was very successful. More recently, single sided deaf patients without tinnitus have been implanted in an effort to improve health-related quality of life. This effort, too, has been successful. This patient population, with one normal hearing ear and one implanted ear, offers an unprecedented window into the sound quality of a cochlear implant. That is, these patients can tell us whether a candidate stimulus presented to the normally hearing ear sounds like the stimulus presented to the cochlear implant. In this talk, we describe our efforts to reproduce the sound of a cochlear implant.

TUESDAY MORNING, 19 MAY 2015 KINGS 3, 9:00 A.M. TO 11:30 A.M.

Session 2aSA

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

Christina J. Naify, Cochair
Acoustics, Naval Research Lab, 4555 Overlook Ave. SW, Washington, DC 20375

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Invited Papers

9:00

2aSA1. Transforming acoustic sensing with high refractive index metamaterials. Miao Yu and Yiongyao Chen (Mech. Eng., Univ. of Maryland, 2181 Glenn Martin Hall, College Park, MD 20742, mmyu@umd.edu)

Acoustic sensors play an important role in many areas, such as safety (e.g., sonar arrays), public health (e.g., ultrasonic imaging), surveillance (e.g., underwater communication and navigation), and industry (e.g., non-destructive damage detection). However, conventional acoustic sensors inevitably suffer from the fundamental pressure detection limit, which hinders the performance of current acoustic sensing technologies. Here, we design high refractive index acoustic metamaterials that have strong wave compression effect to obtain direct amplification of pressure fields in metamaterials. This enables a novel metamaterial enhanced acoustic sensing system that can help overcome the detection limit of conventional systems. Through analytical, numerical, and experimental studies, we demonstrate that this novel acoustic sensing system can help achieve over an order of magnitude enhancement in acoustic pressure detection limit. This will allow the detection of weak acoustic signals below the noise floor of the conventional acoustic sensing systems. This work is expected to impact many fronts that require high performance acoustic sensing.

9:20

2aSA2. Imaging with reconfigurable active acoustic metamaterials. Bogdan Popa and Steven Cummer (Elec. and Comput. Eng., Duke Univ., PO Box 90291, Durham, NC 27708, bogdan.popa@duke.edu)

We present a design procedure to obtain active metamaterial structures that cover a large range of linear and non-linear acoustic responses, and whose flexibility makes it suitable to numerous applications. The presentation will briefly outline the types of material properties enabled, and then focus on the potential of this approach to various imaging applications. The design method is illustrated experimentally using a very thin metamaterial slab whose functionality is set electronically. This ensures that multiple imaging techniques can be demonstrated using the same physical slab by simply swapping the electronic modules that determine the metamaterial behavior. Using this method, we implement a thin acoustic lens whose focal length can be controlled at will. In addition, we show how time reversal imaging techniques can be implemented in real-time and in a continuous manner. Finally, the slab is configured to act as an acoustic multiplexer that takes sound incident from multiple directions and funnels it in exactly one direction.

9:40

2aSA3. Acoustic meta-structures based on periodic acoustic black holes. Fabio Semperlotti and Hongfei Zhu (Aerosp. and Mech. Eng., Univ. of Notre Dame, 374 Fitzpatrick Hall, Notre Dame, IN 46556, fsemperl@nd.edu)

This paper presents a class of three-dimensional fully isotropic acoustic metamaterials that support the development of load-bearing thin-walled structural elements with engineered wave propagation characteristics, i.e. the meta-structures. This class of metamaterials is based on the concept of Acoustic Black Hole (ABH) that is an element able to bend and, eventually, trap acoustic waves. A periodic lattice of ABHs enables wave propagation characteristics comparable with resonant metamaterials but without the need and the fabrication complexity associated with the use of either multi-material or locally resonant inclusions. The dispersion characteristics show the existence of several interesting phenomena such as strong mode coupling, zero group velocity points on the fundamental modes, and Dirac points at the center of the Brillouin zone. Numerical simulations, conducted on a meta-structure made of a thin metal plate with an embedded slab of ABH material, show the existence of a variety of wave propagation effects (e.g., collimation and bi-refraction) in an extended operating range. Such an engineered material can pave the way to the design of thin-walled adaptive structures with fully passive wave management capabilities.

10:00–10:30 Break

Contributed Papers

10:30

2aSA4. Ultrasonic subwavelength focusing above two dimensional membrane metasurface using time reversal. Shane W. Lani, Karim Sabra, and F. Levent Degertekin (Georgia Tech, 801 Ferst Dr., Atlanta, GA 30332, shane.w.lani@gmail.com)

Surface acoustic waves can propagate above immersed membrane arrays, such as of capacitive micromachined ultrasonic transducers (CMUTs). Similar waves on metamaterials and metasurfaces with rigid structures (typically in the kHz range) have been studied and used for tunable band gaps, negative refraction, and subwavelength focusing and imaging. This work demonstrates through simulation and experiments that a 2D membrane array can be used for subwavelength focusing utilizing a time reversal method. The studied structure consisted of the focusing region, which is a dense grid of 7x7 membranes (6.6 MHz resonance) that support the slow surface acoustic waves. Eight additional membranes are located on the same surface outside the focusing region. Subwavelength focusing was performed by using a time reversal method in which the external eight membranes were used as excitation transducers. Modeling results were verified with experiments that were performed with the membranes being actuated electrostatically and the membrane displacements were measured with a laser Doppler vibrometer. Subwavelength focusing (lambda/5) was achieved on the metasurface while a modal decomposition of the spatial focus from an iterative time reversal method was done to illustrate that optimal focusing resolution requires efficient excitation of the mode shapes containing subwavelength features.

10:45

2aSA5. Enhanced acoustic transmission through a slanted grating. Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu)

It is known that an acoustic wave incident on an infinite array of aligned rectangular blocks of a different acoustic material exhibits total transmission if certain conditions are met which relate the unique "intromission" angle of incidence with the geometric and material properties of the slab. This extraordinary acoustic transmission phenomenon holds for any thickness of the slab, making it analogous to a Brewster effect in optics, and is independent of frequency as long as the slab microstructure is sub-wavelength. Here, we show that the enhanced transmission effect can be obtained in a slab with parallel grating elements oriented obliquely to the slab normal. The dependence of the intromission angle θ_i is given explicitly in terms of the orientation angle. Total transmission is also achieved at incidence angle $-\theta_i$ although there is a relative phase shift between the transmitted amplitudes of the $+\theta_i$ and $-\theta_i$ cases. These effects are easily understood when the grating elements are rigid. In particular, any angle of intromission can be obtained by with thin rigid elements by orienting them to the desired value of θ_i.

11:00

2aSA6. Effect of waveguide impedance on radiation in underwater acoustic leaky wave antennas. Christina J. Naify (Acoust., Naval Res. Lab, 4555 Overlook Ave. SW, Washington, DC 20375, christina.naify@nrl.navy.mil), Matthew Guild (National Res. Council, Washington, DC), David C. Calvo, and Gregory J. Orris (Acoust., Naval Res. Lab, Washington, DC)

Leaky wave antennas (LWAs) have been examined for decades as a way to steer electromagnetic waves as indexed by input frequency, enabling rapid video scanning of an environment with a simple compact device. Recently, an LWA device constructed of a rigid waveguide with open shunts was shown to produce a similar steering effect for acoustic waves in air [Naify *et al*., Appl. Phys Lett. **102**, 203508 (2013)]. The shunts serve two purposes, to couple acoustic energy from the waveguide to the surrounding area, as well as control directionality of the radiated wave by changing the impedance, and thus modulating the phase speed of the waves in the waveguide. While this impedance contrast of the waveguide wall relative to that of the surrounding fluid is very large for an air background, when designing a structure for use in water the material of the waveguide can drastically change the directionality of the radiation profile. This study examines the effect of waveguide wall impedance on directionality of an acoustic LWA using both numerical and finite element methods. Three material impedance conditions are examined, including acoustically rigid, and two materials with finite impedance. [Work sponsored by the Office of Naval Research.]

11:15

2aSA7. Effect of air volume fraction and void size on the performance of acoustically thin underwater noise isolation panels. Ashley J. Hicks, Michael R. Haberman, and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Office # N642, Austin, TX 78758, pswilson@arlut.utexas.edu)

We present experimental measurements of the acoustic behavior of underwater sound absorbing and isolating panels composed of three deeply sub-wavelength layers. The panels were constructed with an inner layer of Delrin plastic containing through-thickness circular air-filled holes sandwiched between two rubber layers. Panel isolation efficacy was quanitifed by measuring insertion loss for non-dimensional panel thicknesses, *kd*, ranging from 7.2 x 10^{-3} to 8.5 x 10^{-2}. Finite element numerical simulations will be discussed which indicate that the panel structure mimics a planar encapsulated bubble screen exactly one bubble thick but displays measured performance that is significantly more broadband. Experimental and numerical results indicate that panels with *kd* ranging from 2.8 x 10^{-2} to 7.2 x 10^{-2} can generate a frequency-averaged insertion loss ranging from 5 to 12 dB, dependent on air volume fraction in the panel. The effect of individual air pocket resonance frequency on insertion loss will also be discussed. [Work supported by ONR.]

Session 2aSC

Speech Communication: General Topics in Speech Production (Poster Session)

Brad H. Story, Chair

Speech, Language, and Hearing Sciences, University of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721

Posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to view other posters, contributors of odd-numbered papers will be at their posters from 8:15 a.m. to 9:45 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 11:30 a.m. There will be a 15-minute break from 9:45 a.m. to 10:00 a.m.

Contributed Papers

2aSC1. Experimental analysis of intraglottal flow and geometry. Liran Oren, Sid Khosla, and Ephraim Gutmark (Otolaryngol., Univ. of Cincinnati, PO Box 670528, Cincinnati, OH 45267, orenl@ucmail.uc.edu)

During phonation, skewing of the glottal area and glottal flow refers to phenomenon that occurs when the area or the flow decelerates more rapidly than it accelerates. This skewing is clinically important because it increases the glottal efficiency (defined by the acoustic intensity divided by the subglottal pressure). Current theoretical models predict that skewing of the area is not directly proportional to skewing of the flow due to the affects of vocal tract inertance. Recently, we have developed a method that can measure airflow velocity, the distance between the folds, and the glottal flow rate in excised canine larynx. In the current study, PIV measurements are taken in five excised canine larynges having all the structures above the vocal folds removed. The results show strong correlation between the maximum closing rate at the superior aspect of the folds and the lowest negative pressure that is computed inside the glottis during the closing phase. The results also show that the magnitude of the divergence angle that develops in the glottis during closing is directly proportional to the glottal efficiency. These are important observations because they hold new mechanisms that can increase glottal efficiency without the affects of vocal tract inertance.

2aSC2. Sufficiency of a four-parameter spectral model of the voice source. Jody Kreiman, Bruce R. Gerratt, Rosario Signorello, and Shaghayegh Rastifar (Head and Neck Surgery, UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, jkreiman@ucla.edu)

Our proposed perceptually motivated spectral-domain model of the voice source comprises spectral slopes in four ranges (H1-H2, H2-H4, H4-the harmonic nearest 2 kHz, and that harmonic to the harmonic nearest 5 kHz). Previous studies established the necessity of these parameters by demonstrating that listeners are sensitive to all of these parameters, and that all are needed to adequately model voice quality. The present study examines the sufficiency of the model to quantify voice quality by applying it to a very large set of voices. To determine the range of phenomena for which the source model is valid, 200 natural voices (half male; 50 normal/mildly pathologic, 30 moderately pathologic, 20 severely pathologic) were copy-synthesized using the model, and adequacy of synthesis was assessed in a listening test by comparing the synthetic tokens to the natural targets. Validity of the model as a function of severity of pathology and the particular manner in which the stimuli deviate from normal will be discussed. Sources of mismatches between synthetic and natural stimuli will be identified, and their status with respect to definitions of voice quality will be addressed. [Work supported by NIH.]

2aSC3. Hysteresis in an extension of Titze's surface wave model and contact with phonation onset/offset experiments. Lewis Fulcher (Phys. & Astronomy, Bowling Green State Univ., Ridge Ave., Bowling Green, OH 43403, fulcher@bgsu.edu) and Ronald Scherer (Commun. Sci. and Disord., Bowling Green State Univ., Bowling Green, OH)

Three modifications of Titze's surface wave model are required to incorporate the property of hysteresis. (1) Adding a nonlinearity to the damping term of this model allows it to undergo sustained oscillations above the onset threshold, and thus the model becomes more robust. (2) Adding a nonlinearity to the stiffness term gives the model a frequency dependence that allows contact with the dynamic viscosity measurements carried out by Chan and Rodriguez [J. Acoust. Soc. Am. **124**, 1207–1219 (2008)]. These measurements observed a reduction in the dynamic viscosity as the frequency of oscillation increased. In order to include this property in the surface wave model, (3) the size of the damping constant is reduced whenever the energy of the oscillating vocal fold exceeds a certain value. Thus, the nonlinear properties introduce a dependence of the damping term upon the oscillation history of the vocal fold and set the stage for hysteresis effects. Calculations with the extended surface wave model are compared with the observed onset/offset pressures measured by Titze, Schmidt, and Titze [J. Acoust. Soc. Am. **97**, 3080–3084 (1995)] and by Chan and Titze [J. Acoust. Soc. Am. **119**, 2351–2362 (2006)]. The extended model readily provides an explanation for why the observed phonation offset pressures are always lower than the onset pressures as well as several other trends in the data.

2aSC4. Respiratory-laryngeal coordination in airflow conservation and reduction of respiratory efforts of phonation. Zhaoyan Zhang (Head and Neck Surgery, UCLA School of Medicine, 1000 Veteran Ave., 31-24 Rehab Ctr., Los Angeles, CA 90095, zyzhang@ucla.edu)

Although many previous studies investigated the effects of respiratory-laryngeal coordination during voice production, the importance of respiratory-laryngeal coordination to airflow conservation and reducing respiratory efforts of phonation has not been systematically investigated. In this study, a computational model of the pressure-volume-flow relationship in the respiratory system is developed and coupled to a three-dimensional continuum model of phonation. Simulations at different conditions of the lung initiation volume and laryngeal resistance show that small laryngeal resistance, as observed in vocal fold paralysis or soft voice conditions, leads to large airflow rate out of the lungs and thus below-normal duration of speech between breaths. Although the speech duration can be increased by an increase in either the lung initiation volume or laryngeal resistance, increased laryngeal resistance also reduces the respiratory effort required to maintain a target subglottal pressure. The demand for laryngeal resistance increase in order to maintain a normal breath group duration increases when phonating at a lower lung initiation volume or when a higher subglottal pressure is desired. Finally, the simulations show that tightening vocal fold approximation is more effective than vocal fold stiffening in increasing speech duration between breaths and reducing respiratory effort of phonation. [Work supported by NIH.]

2aSC5. Fundamental frequency movements in one-word imperatives. Sergio Robles-Puente (World Lang., Literatures and Linguist, West Virginia Univ., Chitwood Hall, P.O. Box 6298, Morgantown, WV, Morgantown, WV 26506-6298, seroblespuente@mail.wvu.edu)

F0 contours in Spanish declarative and imperative sentences have traditionally been described as identical and it has not been until recently that several phonetic variables have been identified as markers of imperativity. These include higher overall F0s, early peak alignments in pre-nuclear pitch-accents, upstepped nuclear pitch-accents and marginal use of high boundary tones. Since previous analyses have concentrated on utterances with multiple pitch-accents, not much is know about productions with just one word. This study focuses on one-word utterances given the tendency of imperatives to be short and taking into account that some of the aforementioned phonetic markers cannot be used with a sole pitch-accent. The analysis of 117 declarative sentences and 256 imperatives produced by eight Spanish speakers demonstrated that in cases where declaratives and imperatives share the same contour (L+>H*L%), the latter tend to have higher F0 peaks (Wilcoxon signed-rank test; $p<0.000$). Besides, 32.4% of the imperatives showed F0 contours ending in high boundary tones, a marker not used in declaratives and only marginally found in imperatives with multiple pitch-accents. Current findings suggest that when segmental material limits F0 movements, speakers look for alternative phonetic strategies to distinguish between declaratives and imperatives.

2aSC6. Indirect evidence of perturbation leads to changes in production of voice amplitude and fundamental frequency. Elizabeth D. Casserly (Psych., Trinity College, 300 Summit St., Life Sci. Ctr., Hartford, CT 06106, elizabeth.casserly@trincoll.edu), Lily Talesnick (Neurosci., Trinity College, Hartford, CT), and Nicholas Celestin (Psych., Trinity College, Hartford, CT)

In this study, we explore the juncture of real-time feedback-based changes in speech production and those initiated by top-down knowledge of external factors such as audience characteristics or audience listening conditions. Twenty-four speakers were asked to solve interactive puzzles via remote auditory connection with an experimenter under five signal transmission conditions: normal/baseline, increased signal amplitude, decreased amplitude, +100 cents shift in pitch, and −100 cents shift in pitch. Evidence of these changes was available to talkers only through the speech of the experimenter; no sidetone was introduced in the connection and listening conditions for talkers were otherwise normal. In response to hearing amplitude shifts in their partner's speech, 19/24 talkers significantly altered their mean voice amplitude, while 6/24 altered f0 in response to shifted experimenter vocal pitch (RM-ANOVA, all corrected p's < 0.05). Approximately 30% of responses countered the signal manipulation (e.g., f0 increase in response to low experimenter f0), while the remainder imitated the observed change. This combination of perturbation-like compensation and imitation/accommodation suggests that speakers can interpret transmission circumstances differently, or pursue different speech-related goals, during even very simple, constrained tasks.

2aSC7. Effect of "only" on prosodic focus marking. Elizabeth Chalmers and Jonathan Howell (Linguist, Montclair State Univ., 1 Normal Ave., Montclair, NJ 07043, chalmerse1@mail.montclair.edu)

This study seeks to determine how the inclusion of the focus-sensitive word "only" affects the focus-marking prosody of a sentence. Twenty-one native English speaking adults read 8 question-answer pairs. The questions were written to elicit focus on an object NP (e.g., Who did you see?). In one condition, the answer contained only (e.g., I only saw Mary); in the other condition, "only" was omitted (e.g., I saw Mary.). The recordings were annotated in Praat using forced alignment. We performed linear residualization of F0, amplitude and duration (cf. Breen *et al.* 2009) to remove effects of item and participant. Statistical models of residual pitch and duration on object-NP and verb failed to show any significant differences between the sentences that contain "only" and those not containing "only". These results fail to support theories of utterance-level prominence, which posit a categorical distinction between presentational and contrastive focus (e.g., Katz and Selkirk 2011).

2aSC8. Fathers' use of fundamental frequency in motherese. Mark VanDam, Paul De Palma, and William E. Strong (Speech & Hearing Sci., Washington State Univ., PO Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu)

Studies of motherese or child-directed speech (CDS) have paid scant attention to fathers' speech when talking to children. This study compares mothers' and fathers' use of CDS in terms of fundamental frequency (F_0) production, examining natural speech from a very large database of hundreds of hours of family speech including mothers, fathers, and preschool children. The day-long recordings are collected with specialized recording software and body-worn hardware, then analyzed with automatic speech recognition (ASR) technology (LENA Research Foundation, Boulder, CO). CDS is defined here as speech in a conversational exchange between the parent and child, and adult-directed speech (ADS) is speech between adults or speech in which the child is not a (vocal) participant. Results confirm many reports in the literature of mothers' increased F_0 during CDS. Results fail to show a difference in the F_0 characteristics between fathers' CDS and ADS speech. This shows that children's linguistic experience with fathers is different than with mothers. This result could be useful to improve ASR techniques and better understand the role of usage in natural language acquisition and the role fathers play in the language acquisition process.

2aSC9. Acoustic correlates of creaky voice in English. Sameer ud Dowla Khan, Kara Becker (Linguist, Reed College, 3203 SE Woodstock Boulevard, Portland, OR 97202, skhan@reed.edu), and Lal Zimman (Linguist, Stanford Univ., Stanford, CA)

We compared auditory impressions of creaky voice in English to acoustic measures identified as correlates of contrastive voice qualities in other languages (e.g., Khmer, Chong, Zapotec, Gujarati, Hmong, Trique, and Yi). Sixteen trained linguistics undergraduates listened to the IP-final word "bows" produced five times each by five American English speakers reading the Rainbow Passage, and gave a rating from 0 (no creak) to 5 (very creaky). Results show that stronger auditory impressions of creak are significantly correlated with lower f0, lower cepstral peak prominence (CPP), lower harmonics-to-noise ratios (HNR), and higher subharmonics-to-harmonics ratio (SHR). This suggests that listeners perceive greater creakiness as the voice becomes lower pitched, less periodic, and more audibly interspersed with subharmonic frequencies (i.e., diplophonia). Notably, none of the spectral amplitude measures proposed as acoustic correlates of glottal configurations for creaky voice in other languages (e.g., lower H1-H2 for smaller open quotient, lower H1-A1 for smaller posterior aperture, lower H1-A3 for more abrupt closure, etc.) was significantly correlated with these judgments in any expected direction. Taken together, these results suggest that while listeners consistently use pitch and periodicity as cues to creak, speakers might be varying in their articulatory strategies to achieve those acoustic effects.

2aSC10. Detecting palatalization in spontaneous spoken English. Margaret E. Renwick (Linguist Program, Univ. of Georgia, University of Georgia, 240 Gilbert Hall, Athens, GA 30602, mrenwick@uga.edu) and Caitlin N. Cassidy (Inst. for Artificial Intelligence, Univ. of Georgia, Athens, GA)

We present an analysis of palatalization from /s/ to [ʃ] at word boundaries in UK English. Previous work has considered the effect of lexical frequency (LF) on this phenomenon, but without combining acoustics and spontaneous speech in one study, which we undertake using data gathered from the Audio BNC (http://www.phon.ox.ac.uk/AudioBNC). We analyze 5,259 word pairs in five phonological contexts, comparing the acoustics of test tokens subject to palatalization (e.g., /s/ in *miss you*), to control tokens containing non-alternating [s] (*miss it*) or [ʃ] (*mission, wish it, wish you*). Word and segment boundaries were obtained via forced alignment, but hand-checked. We measured the spectral moments and duration of each fricative; following vowel duration and formant values were also extracted. LF was calculated using the Audio BNC. We find that the spectral center of gravity (CoG) of [s] before [j] (*miss you*) is intermediate ($p<0.01$) between those of [s] and [ʃ]. Furthermore, CoG of test tokens correlates negatively with LF, indicating increased palatalization in high-frequency contexts; no control pair exhibits such correlations. LF predicts CoG even considering speaker gender and other phonetic co-predictors. This supports the view of palatalization as gestural overlap, which increases with LF in casual or fast speech.

2aSC11. Coarticulation and speech stability for three lifespan groups. Alissa Belmont, Stefan A. Frisch, and Nathan Maxfield (Commun. Sci. and Disord., Univ. of South Florida, 4202 E Fowler Ave., PCD1017, Tampa, FL 33620, abelmont@mail.usf.ed)

Using ultrasound recording of tongue movement during speech production, repetitions of kV initial monosyllabic words were measured for nine vowel contexts in three different age groups (children 8–12, young adults 18–35, and older adults 50–65). Coarticulation was measured by average minimum point-to-point distance between tongue traces (Zharkova & Hewlett 2009). Speech stability was quantified by average distance between productions with the same vowel. Coarticulation was quantified by average distance between productions with different vowels. For all three talker groups (n = 11 or greater per group), the measure of coarticulation was greater than the measure of stability, demonstrating coarticulatory differences. Across the lifespan, measures of coarticulation and stability both decreased from group to group. This suggests that speech production becomes more segmental (less coarticulated) and more stable (less variable) continuously throughout the lifespan of speaking experience.

2aSC12. Influences of word frequency and response delay on imitation of visual speech. James W. Dias and Lawrence D. Rosenblum (Dept. of Psych., Univ. of California, 900 University Ave., Riverside, CA 92521, jdias001@ucr.edu)

Human perceivers unconsciously imitate the subtle articulatory characteristics of perceived speech (*phonetic convergence*) during live conversation [e.g., Pardo, 2006] and when shadowing (saying aloud) pre-recorded speech [e.g., Goldinger, 1998]. Perceivers converge along acoustical speech dimensions when shadowing auditory (heard) and visual (lipread) speech [Miller, Sanchez, & Rosenblum, 2010], suggesting the information to which perceivers converge may be similar across sensory modalities. It is known that phonetic convergence to auditory speech is reduced when shadowing high-frequency words and when shadowing responses are delayed. These findings suggest that stored lexical representations and working memory processing time may modulate speech imitation [Goldinger, 1998]. The question arises of whether phonetic convergence to shadowed *visual* speech would demonstrate the same effects of word frequency and shadowing response delay. Phonetic convergence was reduced when shadowing lipread high-frequency words, suggesting lexical information may be similarly accessed by auditory and visual information, consistent with the effects of word frequency on auditory and visual speech comprehension [e.g., Auer, 2002, 2009]. However, where shadowing delay reduced phonetic convergence to auditory speech, it *increased* phonetic convergence to visual speech. The results are discussed with respect to possible differences in processing of auditory and visual speech information across response delays.

2aSC13. Individual differences in phonetic drift by English-speaking learners of Spanish. Marie K. Huffman (Linguist, Stony Brook Univ., SBS S 201, SBS S 201, Stony Brook, NY 11794-4376, marie.huffman@stonybrook.edu) and Katharina Schuhmann (Freie Universität Bozen - Libera Università di Bolzano, Bolzano, Italy)

Second language learning affects L1 pronunciation even in early stages of adult L2 acquisition (e.g., Guion 2003, Chang 2012), an effect sometimes called “phonetic drift.” English voiceless stops assimilate to Korean stops in novice L1 English-L2 Korean learners in Korea, arguably due to English voiceless stops and Korean aspirated stops being linked to the same phonological category (Chang 2012), and the “novelty” of the Korean language input with its longer VOTs (Chang 2014). We tested whether novice L1 English-L2 Spanish learners in the United States also show assimilatory L2-to-L1 effects. Stronger typological and orthographic similarities might lead speakers to equate English and Spanish stops, but phonetic differences—short-lag VOT on voiceless stops and pre-voicing of voiced stops in Spanish—might impede L2-to-L1 assimilation. Monolingual English college-students were recorded every other week for the first 12 weeks of an introductory Spanish course. VOT data show considerable individual differences—some subjects show assimilation, some dissimilation, between English and Spanish stops. This suggests that while orthography and typological similarity may play a role, how a learner’s first and second language interact in novice adult L2 acquisition is also affected by individual factors such as perceptual sensitivity to phonetic differences, sociolinguistics considerations, and possibly speaker strategies for maintaining contrast (e.g., Nielsen 2011).

2aSC14. Auditory feedback perturbation of vowel production: A comparative study of congenitally blind speakers and sighted speakers. Pamela Trudeau-Fisette, Marie Bellavance-Courtemanche, Thomas Granger, Lucile Rapin, Christine Turgeon, and Lucie Ménard (Département de linguistique, Université du Québec à Montréal, C.P. 8888, Succursale Centre-Ville, Montréal, Quebec H3C 3P8, Canada, trudeau-fisette.pamela@courrier.uqam.ca)

Studies with congenitally blind speakers show that visual deprivation yields increased auditory discrimination abilities as well as reduced amplitude of labial movements involved in vowel production, compared with sighted speakers. To further investigate the importance of auditory and visual feedback in speech, a study of auditory perturbation of rounded vowels was conducted in congenitally blind and sighted French speakers. Acoustic and articulatory (electromagnetic articulography) recordings from ten congenitally blind speakers and ten sighted speakers were obtained during the production of the French rounded vowel /ø/. All participants were first asked to produce the vowel repeatedly in a normal condition, i.e., with regular auditory feedback. In the perturbed condition, participants received, in real-time through headsets, an altered version of their speech, in which F2 was gradually increased up to 500 Hz. To compensate for this perturbation, speakers had to enhance lip protrusion and/or tongue retraction. These adaptive maneuvers should have been concurrent with auditory perception abilities. Preliminary results show that congenitally blind speakers gave greater weight to auditory perception than their sighted peers, while compensating differently for the perturbations. These findings support the hypothesis that vision plays a significant role in the implementation of phonological targets.

2aSC15. Vocal imitations of basic auditory features. Guillaume Lemaitre, Ali Jabbari, Olivier Houix, Nicolas Misdariis, and Patrick Susini (IRCAM, IRCAM, 1 Pl. stravinsky, Paris 75004, France, GuillaumeJLemaitre@gmail.com)

We recently showed that vocal imitations are effective descriptions of a variety of sounds (Lemaitre and Rocchesso, 2014). The current study investigated the mechanisms of effective vocal imitations by studying if speakers could accurately reproduce basic auditory features. It focused on four features: pitch, tempo (basic musical features), sharpness, and onset (basic dimensions of timbre). It used two sets of 16 referent sounds (modulated narrow-band noises and pure tones), each crossing two of the four features. Dissimilarity rating experiments and multidimensional scaling analyses confirmed that listeners could easily discriminate the 16 sounds based the four features. Two expert and two lay participants recorded vocal imitations of the 32 sounds. Individual analyses highlighted that participants could reproduce accurately pitch and tempo of the referent sounds (experts being more accurate). There were larger differences of strategy for sharpness and onset. Participants matched the sharpness of the referent sounds either to the frequency of one particular formant or to the overall spectral balance of their voice. Onsets were ignored or imitated with crescendos. Overall, these results show that speakers may not imitate accurately absolute dimensions of timbre, hence suggesting that other features (such as dynamic patterns) may be more effective for sound recognition.

2aSC16. Real-time three-dimensional ultrasound imaging of pre- and post-vocalic liquid consonants in American English. Brandon Rhodes, Kelly Berkson, Kenneth de Jong, and Steven Lulich (Dept. of Linguist., Indiana Univ., 1021 E. Third St., Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu)

Speech sound articulation is typically characterized in the mid-sagittal plane. However, lateral segments, such as the /l/ category in most varieties of English, are usually identified in introductory phonetics courses as exceptions to this rule. On the other hand, many productions of post-vocalic /l/ in varieties spoken in the lower Mid-west U.S. are noted as being non-lateral, involving only a dorsal articulation very different from the canonical

coronal occlusion. Furthermore, a large body of literature indicates multi-constriction articulations, which vary by syllable position, for liquids like American English /l/ and /r/. This research presents results from a study of constriction location and laterality in pre- and post-vocalic /l/ and /r/ using real-time 3D ultrasound images of tongue motion, digitized impressions of the palate, and time-aligned acoustic signals.

2aSC17. Coupling relations underlying complex coordinative patterns in speech production. Leonardo Lancia, Sven Grawunder, and Benjamin Rosenbaum (Linguist, Max Planck Inst. for Evolutionary Anthropology, Deutscher Platz 6, Leipzig 04103, Germany, leonardo_lancia@eva.mpg.de)

In studying linguistic behavior, we are often faced to complex dynamical patterns arising from the highly coordinated activity of many partially autonomous processes. In this work, we apply a new approach aimed at studying abstract coupling relations coordinating the behavior of dynamical systems governed by goal oriented behavior. The approach, based on an original version of recurrence analysis, allows to deal with the principal difficulties of this task, which are mainly related to the heterogeneity, the lack of separability and the lack of stationarity of the processes under study. The method is validated trough simulations of theoretical systems and it is adopted to capture (1) invariant abstract coupling structure underlying systematically varying trajectories of the speech articulators involved in the production of labial and coronal plosive and fricative consonants (produced at slow and fast speech rate by five German speakers and recorded via electromagnetic articulography); (2) systematic differences in the coordination between energy modulations observed at different frequency bands in the acoustic signal and showing the interplay between syllable and stress related processes and its relation to rhythmic differences between typologically different languages (pear story narrations were recorded from five speakers per language in German, Italian, French, English, and Polish).

2aSC18. Kinematic properties of concurrently recorded speech and body gestures and their relationship to prosodic structure. Jelena Krivokapic (Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109-1220, jelenak@umich.edu), Mark Tiede (Haskins Labs., New Haven, CT), and Martha E. Tyrone (Dept. of Commun. Sci. and Disord., Long Island Univ. Brooklyn, Brooklyn, NY)

Previous research has shown that body gesturing, such as hand and torso movements, is related to prosodic structure (e.g., Esteve-Gibert & Prieto 2013). However, the details of the relationship are not well understood and specifically the effects of prosodic structure on gesturing duration have received little attention. We present an experiment investigating the effects of stress and prosodic boundaries on the temporal properties of body gesturing and gestures of the vocal tract. Two speakers read 6 repetitions of 12 sentences examining the effect of boundary (no boundary, ip boundary, IP boundary), stress (no stress, stress) and their interaction, phrase finally and phrase initially (144 sentences total). They were observed through concurrent recordings of speech audio, vocal tract gestures using electromagnetic articulometry, and body gesturing using Vicon motion capture. We test the hypothesis that manual movements lengthen under prominence and at boundaries and that the lengthening at boundaries is cumulative, parallel to the behavior of gestures of the vocal tract. Analyzing speech prosody and body gesturing in this manner will allow us to investigate whether prosodic control extends to body gesturing and to identify coordinative patterns that are relevant to fields such as language pedagogy and speech pathology. [Work supported by NIH.]

2aSC19. Articulation of vowel height in Taiwanese Vowels: An EMA study. Yuehchin Chang, Yi-cheng Chen, and Feng-fan Hsieh (Inst. of Linguist, National Tsing Hua Univ., 101, sec. II, Kuang Fu Rd., Hsinchu, Taiwan 300, Taiwan, ycchang@mx.nthu.edu.tw)

Whalen *et al.*'s (2010) ultrasound study has suggested that constriction degree "may be the best descriptor for height" in American English front vowels {i, ɪ, e, ɛ}. The present study took a further step, investigating the case of height and backness distinction in Taiwanese front and back vowels. Stimuli consisted in /CV/ sequences with V corresponding to one of the six vowels {i, e, a, ɔ, ɤ, u} (Southern variety). Five adult speakers were asked to repeat each sequence six times in random order. The articulatory data were collected using the Carstens AG500 EMA system. Our results show that vowel height and backness in Taiwanese vowels can be reliably differentiated as follows: (1) for front and central vowels, height is primarily distinguished by Tongue Body Constriction Degree (TBCD) (i>e>a) (less robustly by Tongue Tip Constriction Degree), whereas height distinction of back vowels is made in Tongue Dorsum Constriction Degree (TDCD) (u>ɤ>ɔ), (2) backness can be distinguished by Tongue Tip and Tongue Body Constriction Location. Therefore, Whalen *et al.*'s (2010) claim can be extended to back vowels as well (at least in Taiwanese). Furthermore, our results are compatible with Ladefoged's (2001) "holistic" view of vowel height as long as his "highest point of the tongue" can be "locally" interpreted: TBCD for height of front and central vowels and TDCD for back vowels. In sum, the present study suggests that distinction in vowel height be relativized to different gestural constriction degree.

2aSC20. The effect of palate morphology on lingual shape during vowel production. Jennell Vick, Michelle L. Foye (Psychol. Sci., Case Western Reserve Univ., 11635 Euclid Ave., Cleveland, OH 44106, jennell@case.edu), Nolan Schreiber, and Gregory S. Lee (School of Eng., Case Western Reserve Univ., Cleveland, OH)

A previous report described significant differences between males and females in the shape of the tongue during the production of vowels (Vick *et al.*, 2014). The purpose of the present study was to understand the influence of palate morphology on these measured differences in tongue shape. This study examined tongue movements in consonant-vowel-consonant sequences drawn from words in phrases as produced by 32 participants, including 12 adults (6m, 6f) and 20 children (2m and 2f at each age of 11, 12, 13, 14, and 15 years). Movements of four points on the tongue were tracked at 100 Hz using the Wave Electromagnetic Speech Research System (NDI, Waterloo, ON, CA). Palate curvature measures were derived from a hard palate trace made with a 6dof probe sensor. The phrases produced included the vowels /i/, /I/, /æ/, and /u/ in words (i.e., "see," sit," cat," and "zoo"). The horizontal curvature of the tongue was calculated at the trajectory speed minimum associated with the vowel production using a least-squares quadratic fit of the positional coordinates of the four points on the tongue. A regression model was used to determine the variance in tongue shape explained by palate morphology in the 32 talkers.

2aSC21. An articulatory model of the velum developed from cineradiographic data. Yves Laprie (LORIA/CNRS, 615 rue du jardin botanique, Villers-lès-Nancy 54600, France, yves.laprie@loria.fr)

The objective of this work is to develop an articulatory model which comprises the velum so as to be able to synthesize nasal vowels and consonants in French. Unlike one exception of a velum model developed from static 3D MRI images there is no model derived from articulatory data that can be used within a vocal tract articulatory model. Our model is derived from X-ray films of the DOCVACIM database. The velum contour has been tracked semi-automatically and then carefully checked and corrected from 1050 mid-sagittal images of the vocal tract. Principal component analysis has been applied onto these contours. The first component represents more than 50% of the variance and controls the velum opening. The next two components capture 20 and 12 % of the variance, respectively, and explain the deformation of the velum between open and close configurations. In parallel a series of MRI images of nasal vowels and consonant of French was analyzed. The velum opening roughly has a shape of a bean, whose length is almost constant for one speaker and width is given by the 2D articulatory model. Together with an evaluation of the nasal cavities this model can be used for articulatory synthesis.

2aSC22. A task dynamic approach to the coda-voicing effect on vowel duration. Robert Hagiwara (Dept. of Linguist., Univ. of MB, Winnipeg, MB R3T 5V5, Canada, robh@umanitoba.ca)

The coda-voicing effect on vowel duration (CVE), in which vowels are longer with voiced codas and shorter with voiceless, is well attested. However, the precise mechanism of this relationship is not well studied.

2a TUE. AM

Task-dynamic models such as articulatory phonology (e.g., Browman & Goldstein, 1986) offer two ways to affect the duration of a vowel in a CVC syllable: the relative *phasing* of the onset, vowel, and coda gestures (when they start relative to one another), and the relative *stiffness* of an individual gesture (roughly how quickly it is executed). Onosson (2010) argued that for /ai/, the principal mechanism of Canadian Raising (CR) was vowel shortening via increased overlap of the onset gesture and the vowel, rather than vowel-coda phasing or stiffening the vowel gesture. This study investigates whether this explanation holds generally for 15 vowels in Canadian English, using controlled wordlist-style data (originally from Hagiwara, 2006). Preliminary investigation suggests that onset-phasing may hold for the diphthongs, but does not adequately characterize CVE across the vowels. The observable mechanism(s) of CVE will be discussed, as well as implications for CVE and related effects (e.g., CR) across dialects/languages generally.

TUESDAY MORNING, 19 MAY 2015 — BALLROOM 1, 8:00 A.M. TO 11:55 A.M.

Session 2aSP

Signal Processing in Acoustics, Speech Communication and Psychological and Physiological Acoustics: Characteristic Spectral and Temporal Patterns in Speech Signals

Xiao Perdereau, Cochair
Burgundy University, 9, Av. A. Savary, BP 47870, Dijon 21078, France

Daniel Fogerty, Cochair
Communication Sciences and Disorders, University of South Carolina, 1621 Greene St., Columbia, SC 29208

Chair's Introduction—8:00

Invited Papers

8:05

2aSP1. The contribution of temporal modulations to speech intelligibility: Some thoughts about the speech envelope. Ken W. Grant (Audiol. and Speech Ctr., Walter Reed National Military Med. Ctr., 301 Hamilton Ave., Silver Spring, MD 20901, ken.w.grant@gmail.com)

Rosen [Phil. Trans. R. Soc. Lond. B **336**, 367–373 (1992)] laid out an extremely useful framework for discussing temporal information in speech. Three broad categories were defined: temporal envelope (2–50 Hz), periodicity (50–500 Hz), and temporal fine-structure (600–10,000 Hz). The slower modulations associated with the speech envelope convey segmental cues for voicing and manner-of-articulation, and prosodic cues for syllabification and stress. These cues are manifest in the attack and fall times as well as the overall duration of the envelope segments. The mid-rate fluctuations associated with periodicity convey manner cues (periodic vs aperiodic) as well as voice pitch (an essential cue for intonation and stress). The faster rate modulations associated with temporal fine-structure (modulations within each pitch period) convey place-of-articulation cues and spectral-shape cues for manner-of-articulation and voicing. Studies that try to separate the effects of each of these temporal categories on speech perception often use signal processing strategies that filter out the temporal modulations of interest and present this information without periodicity or temporal fine-structure to listeners. The processing details of how the temporal envelope is extracted are critical to whether cues more readily associated with periodicity or fine-structure are inadvertently presented to listeners. Examples will be provided.

8:25

2aSP2. Developmental time course of auditory perception of modulation speech cues. Christian Lorenzi (Institut d'Etude de la Cognition, Ecole normale superieure, 29 rue d'Ulm, Paris, Ile de France 75005, France, christian.lorenzi@ens.fr)

Speech contains strong amplitude modulation (AM) and frequency modulation (FM) cues, which are commonly assumed to play an important role in speech identification for adults. We will review recent studies aiming to characterize the development of auditory perception of AM and FM speech cues for 6 and/or 10-month-old infants learning French or Mandarin [e.g., Cabrera, L., Tsao, F.-M., Gnansia, D., Bertoncini, J., & Lorenzi C. (2014), J. Acoust. Soc. Am. **136**, 877–882; Cabrera, L., Bertoncini, J., & Lorenzi, C. (2013), J. Speech, Lang., Hearing Res. **56**, 1733–1744]. These studies were based on vocoders, which are analysis and synthesis systems designed to manipulate the modulation components of speech sounds in a given number of frequency bands. Overall, the results suggest that: (i) the auditory processing of AM and FM speech cues is "functional" by 6 months, (ii) the auditory processing of the AM and FM cues is fine-tuned by language exposure between 6 and 10 months. These findings may help improving current models of modulation processing that do not take into account the plasticity of the auditory and speech-processing system.

8:45

2aSP3. Contributions of the temporal envelope in adverse listening conditions. Daniel Fogerty (Commun. Sci. and Disord., Univ. of South Carolina, COMD, Keenan Bldg., Ste. 300, Columbia, SC 29208, fogerty@sc.edu)

Amplitude modulations of the temporal envelope provide important cues for intelligibility, particularly in quiet listening environments. However, the speech understanding of listeners based on this temporal speech signal is often impaired when temporal interruptions or maskers interfere with the transmission of this important information. Performance in these adverse listening conditions is variable, based, in part, upon the properties of the interfering masker and the degree to which the modulation properties of the target speech signal are preserved. In some cases, listeners may be required to shift perceptual weighting to alternate, preserved modulation cues in order to maximize intelligibility of the speech signal. However, modulation properties of the speech signal are related to specific speech events. Therefore, different modulation properties are likely to contribute diverse information for speech understanding. The ability to access and process temporal modulation properties of the speech signal that unfold over different timescales is essential for maximizing speech intelligibility in everyday listening conditions. [Work supported by NIH/NIDCD and ASHA.]

9:05

2aSP4. Segmental contribution to the intelligibility of noise-suppressed speech. Fei Chen (Dept. of Elec. and Electron. Eng., South Univ. of Sci. and Technol. of China, Shenzhen 518055, China, fchen@sustc.edu.cn)

Many studies suggested that most speech enhancement algorithms do not improve the subjective intelligibility scores of noise-suppressed speech when presented to normal-hearing listeners. Nevertheless, the reason for lacking intelligibility improvement is still unclear. This study assessed the segmental contribution to the intelligibility of noise-suppressed speech. Mandarin sentences were corrupted by steady-state speech-spectrum shaped noise and multi-talker babble. Stimuli were synthesized by using noise-suppressed speech processed by three types of speech enhancement algorithms and a noise-replacement paradigm. The noise-replacement paradigm preserved selected speech segments (i.e., vowel-only and vowel-plus-consonant-onset), and replaced the rest with noise. Listening experiments were performed to collect the intelligibility scores from normal-hearing listeners. Experimental results showed that the selectively synthesized stimuli (i.e., vowel-only and vowel-plus-consonant-onset) from noise-suppressed speech may be more intelligible than those from noise-corrupted speech. However, this benefit was only observed when the speech signal was corrupted by speech-spectrum shaped noise but not by babble masker. This work suggested that the segmental distortion caused by speech enhancement algorithms may affect the information integration from noise-suppressed speech, and high-level top-down processing may account for the deficiency of the present speech enhancement algorithms for improving subjective intelligibility performance.

9:25

2aSP5. Using speech intelligibility metrics to predict the performance of hearing-impaired listeners. Karen Payton (Elec. & Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747-2300, kpayton@umassd.edu)

When fitting a hearing aid to a hearing-impaired listener, it would be very beneficial for an audiologist to be able to use an objective metric to determine the best signal processing algorithm to compensate for the user's loss and improve the user's understanding of speech in degraded acoustic environments. Several metrics have been considered to predict impaired listeners' speech recognition, including modified versions of the Speech Intelligibility Index (SII) [e.g., Ricketts, Ear Hear. **17**, 124–132 (1996)] and the Speech Transmission Index (STI) [e.g., Humes *et al*., J. Speech Hear. Res. **29**, 447–462 (1986)]. For impaired listeners using linear amplification hearing aids, there are metrics that work fairly well. For listeners using hearing aids that include nonlinear processing, such as amplitude compression, results have not been as successful. This talk will review the current status of metric capabilities, approaches that have been proposed to mitigate metric limitations and which metric features appear most promising for future research.

9:45

2aSP6. A novel method to measure the resonances of the vocal tract. Bertrand Delvaux and David Howard (Dept. of Electronics, Univ. of York, Heslington, York, York YO10 5DD, United Kingdom, bertrand.delvaux@gmail.com)

In speech/singing, knowledge of the frequencies of the resonances of the vocal tract gives access to the vowel type (lower resonances) and the voice timbre (higher resonances). It is therefore crucial to be able to measure accurately these resonant frequencies directly. Several approaches have been developed such as glottal excitation, excitation at the lips, or medical imaging. More accurate measurements of the vocal tract resonances have emerged from physical modeling in recent years but a simple-to-use *in vivo* measurement tool would be much more valuable to give feedback to the singer/actor and their teacher(s), as well as voice therapists, language teachers, speech scientists, phoneticians, and linguists. In this article, we suggest a novel method using a sinesweep to measure the resonant frequencies of the vocal tract simultaneously with the voice spectrum.

10:05–10:20 Break

10:20

2aSP7. An acoustic phonetic study on the production accuracy of lexical stress contrasts in Arabic and Mandarin-accented English. Paul R. Keyworth (Appl. Linguist, Saint Cloud State Univ., 3710 W. Saint Germain St., Apt. #234, Saint Cloud, MN 56301-7319, kepa1104@stcloudstate.edu)

The researcher explored the socio-morphophonetic characteristics of English as a second language lexical prosodic competence. One hundred participants from Midwest USA (29), Saudi Arabia (38), and China (33) were recorded producing English lexical stress (ELS) in tokens containing seven different stress-moving suffixes—[-ic], [-ical], [-ity], [-ian], [-ify], [-ial], and [-ious]. Fundamental frequency, duration, and intensity productions were analyzed using *Praat*. In total, 2125 vowels in 800 spectrograms were analyzed (excluding stress placement and pronunciation errors). Statistical sampling techniques were used to evaluate acquisition of accurate ELS production versus native language and language proficiency. Speech samples of native-speakers were analyzed to provide norm values for cross-

reference and to provide insights into the proposed *Salience Hierarchy of the Acoustic Correlates of Stress* (SHACS). The results support the notion that a SHACS does exist in the L1 sound system and that native-like command *is* attainable for English language learners through increased study/L2 input. Saudi English speakers, who often do not fully reduce vowels, produced durational contrasts more accurately when they had studied English for longer. Similarly, proficient Chinese learners of English seemed to be able to overcome negative transfer from their tonal system as they produced pitch in a more native-like manner.

10:40

2aSP8. Investigating the effectiveness of Globalvoice CALL software in native Japanese English speech. Hiroyuki Obari (Economics, Aoyama Gakuin Univ., 2-12-21, Nakatakatsu, Tsuchiura, Ibaraki 300-0815, Japan, obari119@gmail.com), Hiroaki Kojima (National Inst. of Adv. Industrial Sci. and Technol. (AIST), Tsukuba, Ibaraki, Japan), and Shuichi Itahashi (Univ. of Tsukuba, Aobaku, Sendai, Japan)

Many variables exist in the evaluation of the speech of non-native speakers of English. In previous research, rhythmic accent and pauses are more salient than segmental features in English utterances to make one's speech more intelligible. In this regard, prosodic features such as intonation and rhythm are crucial in comprehensible speech. One of the main goals of English education in Japan is to help Japanese students speak English more intelligibly in order to be more clearly understood while taking part in international communication. In this talk, several key parameters such as speech duration, speech power, and F0 (pitch), are all introduced to help determine to what extent Japanese students are able to improve their English pronunciation through the use of Globalvoice CALL software. Globalvoice CALL enables students to input English words or sentences for the purpose of practicing their pronunciation and prosody in order to reduce their Japanese-accented speech. The purpose of this paper is to investigate the effectiveness of Globalvoice CALL software to improve the English pronunciation of Japanese EFL learners. Student productions of English sentences were analyzed with regard to speech duration, speech power, and F0 (pitch) to determine how much progress was attained in their English pronunciation and overall proficiency. The results showed improvement in acoustic properties between the pre- and post-recorded readings, indicating the software helped the students to improve their English pronunciation.

11:00

2aSP9. Contribution of the acoustic cues to the non-native accent. Yves Laprie (LORIA/CNRS, 615 rue du jardin botanique, Villers-lès-Nancy 54600, France, yves.laprie@loria.fr)

This communication concerns the contribution of acoustic cues to the perception of non-native accents and presents some solutions to help learners to master the acoustic cues of the foreign language. Elementary acoustic cues (F0 frequency, formants…) will be presented by using copy synthesis used to pilot the Klatt formant synthesizer. This enables their perceptual contribution and that of more complex acoustic cues, which play a major role in the identification of speech sounds, to be evaluated easily. The non-native accent is mainly due to the:—realization of incorrect tones, stress patterns, and more generally prosody,—use of the mother tongue phonemes instead of those of the second language,—use of inappropriate acoustic cues to realize a phonetic feature. This is often the case with the voiced/unvoiced feature which covers a range of acoustic cues. These deviations depend on the phonetic specificities of the mother tongue with respect to those of the second language. The first aspect of corrections is the automatic diagnosis of non-native deviations. Learners can be made aware of them, trained to learn new phonetic contrasts, or guided by providing them with appropriate articulatory information, or acoustic feedback which is particularly efficient for prosody.

11:20

2aSP10. Speech rhythm processing. Xiao Perdereau (Laboraoire Interdisciplinaire Carnot de Bourgogne UMR 6303 CNRS, 9, Av. A. Savary, BP 47870, Dijon 21078, France, amiscec@free.fr)

Speech rhythm is one of the main prosodic features of spoken languages. The quantitative research data on this topic in the literature are not numerous may be due to the confusing divergence in this concept definition. In order to render clear this concept, we investigated a speech segment in Mandarin. Due to its delicate temporal structure, it leads easily to the semantic ambiguities, well known in speech recognition. In order to understand how human manage the speech meaning in a natural language processing, we designed a series of lexical duration and pause interval modulations based on the speech segment for production. By analyzing the resultant acoustic patterns, we observed two types of temporal grouping. The type 1 is context dependent, the time range has no regularity, the modulations suffered from semantic ambiguities. The type 2 includes the stable patterns presenting temporal regularities. We consider the temporal structures in this type as speech rhythm. This definition corresponds to the articulatory periodicity that produces the consonant and vowel alternations. They are context independent and reproducible.

Contributed Paper

11:40

2aSP11. Using automatic speech recognition to identify pediatric speech errors. Roozbeh Sadeghian (Dept. of Elec. and Comput. Eng., State Univ. of New York at Binghamton, 4400 Vestal Parkway East, Binghamton, NY 13902, rsadegh1@binghamton.edu), Madhavi Ratnagiri (Speech Res. Lab., Nemours Biomedical Res., Wilmington, DE), Stephen A. Zahorian (Dept. of Elec. and Comput. Eng., State Univ. of New York at Binghamton, Binghamton, NY), and H. Timothy Bunnell (Speech Res. Lab., Nemours Biomedical Res., Wilmington, DE)

Speech delay is a childhood language problem that might resolve without intervention, but might alternatively presage continued speech and language deficits. Thus, early detection through screening might help to identify children for whom intervention is warranted. The goal of this work is to develop Automatic Speech Recognition (ASR) methods to partially automate screening for speech delay in young children. Speech data were recorded from typically developing and speech delayed children (N = 63) aged 6 to 9 years old during administration of the Goldman Fristoe Test of Articulation (GFTA). Monophone Hidden Markov Model (HMM) acoustic models were trained on speech data obtained from 207 typically developing children in the same age range. These training data consisted of a total of about 18,000 single-word utterances. The HMMs were then used to develop an utterance verification system to distinguish correct versus error productions. Several variations of the recognition strategy, feature extraction, and scoring methods were investigated. The best overall ASR result for distinguishing normal versus abnormal speech is approximately 86%. It is hypothesized that the ASR methods could approach the level of accuracy of speech therapists for this task (agreement among multiple therapists is over 95%), but a much larger database may be needed.

Session 2aUW

Underwater Acoustics: Historical Perspectives on the Origins of Underwater Acoustics I

David Bradley, Cochair
Penn State University, PO Box 30, State College, PA 16870

Thomas G. Muir, Cochair
Applied Research Laboratories, University of Texas at Austin, P/O. Box 8029, Austin, TX 78713

Chair's Introduction—8:00

Invited Papers

8:10

2aUW1. Paul Langevin's contributions to the development of underwater acoustics. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Paul Langevin made significant contributions to the understanding of piezoelectricity and the development of piezoceramics materials. For instance, Professor Langevin's invented the quartz sandwich transducer in 1917 for underwater sound transmission. He subsequently used this to develop the first underwater Sonar for submarine detection during World War I, and his work was extensively used the French and American Navy. After world war I, SONAR devices he helped developed were used in several French ocean-liners. We will review Paul Langevin's most noteworthy contributions to the development of underwater acoustics.

8:30

2aUW2. Harvey C. Hayes: First superintendent of the Sound Division at the Naval Research Laboratory, Washington, D.C. Fred T. Erskine (none, 135 Kensington Dr., Littlestown, PA 17340-9767, kx3z@aol.com)

Harvey Cornelius Hayes, physicist, had a long tenure (1923–1947) as the first Superintendent of the Sound Division (later renamed Acoustics Division) at the Naval Research Laboratory (NRL), Washington, D.C. In the period prior to World War II, Hayes led a small group of only five to eight researchers that was devoted to developing active (echo-ranging) sonar and improved passive (listening) sonar for the navy's surface ships and submarines. They developed a tunable type of sonar that found widespread use in World War II. They conducted field experiments to take detailed measurements on the propagation of sound in oceanic acoustic ducts. They developed techniques for silencing "singing" propellers in navy ships and aircraft carriers by sharpening propeller edges. During World War II the Sound Division expanded in size about twentyfold and NRL researchers conducted numerous experiments to address navy concerns regarding sonar performance. Sound Division researchers made significant advances including the development of communications equipment for divers, torpedo countermeasures, the development of streamlined sonar domes for ships, control and stabilization of sonar transducers, methods for localization of a submerged target beyond the resolution of the sonar beam, and coordination of sonar systems with fire-control systems. In the period following World War II, Hayes initiated new efforts on shock and vibration, crystal development, and physiological acoustics.

8:50

2aUW3. The Submarine Signal Company. Thomas R. Howarth (NAVSEA Div. Newport, 1176 Howell St., B1346 R404A, Newport, RI 02841, thomas.howarth@navy.mil)

The Submarine Signal Company (SSC) was established in 1901 in Boston, MA, and was the first commercial enterprise organized to conduct underwater sound research and to develop equipment to be used for increasing safety of navigation. The initial product line (prior to 1914) included underwater bells for shore based stations, buoys, and lightships, as well as encased microphones for sound detection on the ships. The bells were used for navigational and hazard warning purposes. In April 1912, future SSC President H. J. W. Fay had a chance meeting with Professor Reginald Fessenden at a Boston railroad station. This meeting led to an invitation for a Fessenden visit to SSC where upon Reginald so impressed company managers and engineers that SSC hired Fessenden consulting services to redesign their hydrophones. Based on this and later contributions to underwater sound products, H. J. W. Fay and R. A. Fessenden are two of the five named pioneers of the ASA award known as "The Pioneers of Underwater Acoustics Medal". This presentation will discuss the formation and product lines of SSC through their formation till 1946 when they were merged with Raytheon Company.

9:10

2aUW4. The naval science of Albert Beaumont Wood, O.B.E., D.Sc. Michael J. Buckingham (SIO, UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

A. B. Wood's career in Naval scientific research spanned a period of nearly 50 years, from the early days of the first World War to the time of his death in 1964. After graduating from Manchester University with a first class honors degree in Physics he seemed destined for a career of distinction in academia. But, like many young men at the time, when the U.K. was becoming deeply embroiled in war, he felt dissatisfied with the cloistered academic life and instead became one of the first two physicists to receive an official appointment with the Admiralty under the newly formed "Board of Invention and Research". Thus, the Royal Navy Scientific Service was born and Dr. Wood began his celebrated research into the propagation of sound underwater, about which little was known at the time. Many of his technical achievements were made at the Admiralty Research Laboratory, Teddington and, shortly before his death, he spent a year at the U.S. Naval Electronics Laboratory, as it was then called, in San Diego. He was awarded the Pioneer of Underwater Acoustics medal by the Acoustical Society of America and his Text Book of Sound is still a standard work on the subject.

9:30

2aUW5. Leonid Brekhovskikh and his lasting impact on underwater acoustics. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., 325 Broadway, Mail Code R/PSD99, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

This paper will discuss Brekhovskikh's groundbreaking contributions into physical oceanography and the theory of wave propagation, scattering, and diffraction from the prospective offered by current developments in underwater acoustics. In hindsight, three major contributions stand out in Brekhovskikh's legacy, which highlight different facets of his personality and his talents. For many decades, ocean acoustics was Leonid's passion. He developed and promoted an approach to underwater acoustics, which embraced the ocean's complexity and centered on devising techniques to reliably characterize the uncertain environment by acoustic means. The approach led to creation of an informal but rather productive and enduring "Brekhovskikh's school." Leonid worked only briefly on wave scattering by rough surfaces. However, his findings, which became known as the tangent plane approximation, revolutionized the field. Universally recognized are Brekhovskikh's systematic theoretical studies of underwater sound propagation, which are summarized in his celebrated book *Waves in Layered Media*. The theory includes spectral representations of wave fields, normal mode theory for open waveguides, and a clear treatment of diffraction phenomena attendant to caustics, lateral waves, and reflection of wave beams and pulses. The book charted the ways forward which have been and are followed by numerous researchers around the globe.

9:50

2aUW6. R. J. Urick—Introduction of the sonar equations. David Bradley (Penn State Univ., PO Box 30, State College, PA 16870, dlb25@psu.edu)

The Sonar Equations were developed during the Second World War to calculate the maximum ranges of sonar equipment. They were formally discussed in a series of reports published by the Office of Naval Research in the mid 1950's entitled "A Summary of Acoustic Data" authored by R. J. Urick and A. W. Pryce. Later codified in Urick's Book, Principles of Underwater Sound, they are the primary tool for design, development, and testing of undersea acoustic hardware. They also provide a useful means for estimating expected values of marine physical properties (e.g., ambient noise, transmission loss) or, conversely, a sanity check on data collected at sea. The presentation is focused on the history of the sonar equations, their early use and transition to modern form and employment.

10:10–10:30 Break

10:30

2aUW7. The University of California Division of War Research and the Marine Physical Laboratory. W A. Kuperman (Scripps Inst. of Oceanogr., Univ. of California, San Diego, Marine Physical Lab., La Jolla, CA 92093-0238, wkuperman@ucsd.edu)

In 1941, motivated by the large-scale destruction of ships by submarines in World War II, three university-operated laboratories were established: Columbia University Division of War Research, Harvard Underwater Sound Laboratory, and The University of California Division of War Research (UCDWR). UCDWR was led initially by Vern Knudsen of the University of California, Los Angeles, with senior staff recruited from academic institutions across the country. After the war, UCDWR was dissolved with the The Navy Electronics Laboratory absorbing most of the UCDWR employees. Further, by agreement between the Navy and the Regents of the University of California, the Marine Physical Laboratory (MPL) was formed with an initial scientific staff of five people from UCDWR with Carl Eckart as its Director. We review this transition period and MPL's subsequent trajectory within the University of California system to its ultimate venue in the Scripps Institution of Oceanography.

10:50

2aUW8. Contributions to the development of underwater acoustics at the Harvard Underwater Sound Laboratory. Frederick M. Pestorius (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78759, pestoriu@arlut.utexas.edu) and David T. Blackstock (Appl. Res. Labs. and Mech. Eng. Dept., Univ. of Texas at Austin, Austin, TX)

The Harvard Underwater Sound Laboratory (HUSL) began in June 1941, six months before Pearl Harbor, and closed on January 31, 1946. HUSL was directed throughout its existence by (then) Assoc. Prof. Frederick V. Hunt, who incidentally coined the acronym sonar. This paper traces several contributions HUSL made to undersea warfare through exploitation of underwater acoustics. Especially significant were developments by HUSL in sonar and underwater ordnance (torpedoes). Both developments were supported by numerous subprojects on a wide range of applications such as reverberation suppression, Doppler enhancement, and torpedo transducer design. Particular attention is paid to work on air-dropped and submarine-launched acoustic homing torpedoes. HUSL's far-reaching influence continues to be evident in today's sonar and torpedo systems. Closure of HUSL spawned Penn State's Ordnance Research Laboratory (now Applied Research Laboratory) and enriched sonar development at the US Navy Underwater Sound Laboratory, New London.

Moreover, HUSL personnel returning to the University of Texas soon founded the Defense Research Laboratory (now Applied Research Laboratories) at Texas. We pay particular homage to living HUSL alumnus A. Wilson Nolle, who, along with our recently departed colleague, Reuben Wallace, and 12 other deceased alumni, had come to HUSL from the University of Texas.

11:10

2aUW9. Underwater acoustics research at the Woods Hole Oceanographic Institution, 1930–1960. James Lynch, Arthur Newhall, and Robert Frosch (Woods Hole Oceanographic, MS # 11, Bigelow 203, Woods Hole Oceanographic, Woods Hole, MA 02543, jlynch@whoi.edu)

The Woods Hole Oceanographic Institution (WHOI) was founded in 1930, and throughout its history has had a strong involvement in research into the science and applications of sound in the ocean. In terms of a brief history, three eras stand out: (1) pre-WWII, (2) WWII, and (3) the postwar years. Though the most colorful pictures and stories come from the war years, the other two eras also hold much interest. Many of the names associated with research at WHOI also commonly appear associated with other laboratories, and the interplay between various institutions is a fascinating part of the history. Personal reminisces, technical report details, photos, audio, and even some video records will be presented, courtesy of the WHOI Archives.

11:30

2aUW10. Columbia Division of War Research and the work of Ewing, Worzel, and Pekeris. D. K. Knobles, Evan K. Westwood, and Thomas G. Muir (Appl. Res. Labs., Univ. of Texas at Austin, P.O Box 8029, Austin, TX 78713, knobles@arlut.utexas.edu)

Three University laboratories were established in WWII to conduct underwater acoustics research, in response to widespread attacks on allied shipping by German submarines; the Columbia University Division of War Research, Harvard Underwater Sound Laboratory, and the University of California Division of War Research, and the last two of these are being discussed by other speakers in this session. During the war, outstanding work was done at Columbia by Maurice Ewing, J. Lamar Worzel, and C. L. Pekeris, which has been summarized in the book, Propagation of Sound in the Ocean. Experiments utilizing explosive sources were done at sea in 1943 and 1944, in both shallow and deep water, off the east coast of the United States, with the USS Saluda (IX87) and other vessels, in cooperation with the Naval Ordnance Laboratory, Woods Hole Oceanographic Institution, and the U.S. Navy Underwater Sound Laboratory. Pekeris is noted for his wave theoretic analysis of the shallow water propagation results and is credited with being the first to apply normal mode theory to shallow water waveguides. Ewing and Worzel are noted for their development of some of the first long range ocean acoustics experiments, which continue to this day in similar context and importance. Some highlights of this work are presented to illustrate the beginnings of long range underwater acoustic propagation research in the United States.

Meeting of the Standards Committee Plenary Group

to be held jointly with the meetings of the

ANSI-Accredited U.S. Technical Advisory Groups (TAGs) for:
ISO/TC 43, Acoustics,
ISO/TC 43/SC 1, Noise,
ISO/TC 43/SC 3, Underwater acoustics
ISO/TC 108, Mechanical vibration, shock and condition monitoring,
ISO/TC 108/SC 2, Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures,
ISO/TC 108/SC 3, Use and calibration of vibration and shock measuring instruments,
ISO/TC 108/SC 4, Human exposure to mechanical vibration and shock,
ISO/TC 108/SC 5, Condition monitoring and diagnostics of machine systems, and
IEC/TC 29, Electroacoustics

P.D. Schomer, Chair, U.S. Technical Advisory Group for ISO/TC 43 Acoustics and ISO/TC 43/SC 1 Noise
Schomer and Associates, 2117 Robert Drive, Champaign, IL 61821

M.A. Bahtiarian, Chair, U.S. Technical Advisory Group for ISO/TC 43/SC 3 Underwater acoustics
Noise Control Engineering, Inc., 799 Middlesex Turnpike, Billerica, MA 01821

W. Madigosky, Chair of the U.S. Technical Advisory Group for ISO/TC 108 Mechanical vibration, shock and condition monitoring
MTECH, 10754 Kinloch Road, Silver Spring, MD 20903

M. L'vov, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 2 Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles, and structures
Siemens Energy, Inc., 5101 Westinghouse Blvd., Charlotte, NC 28273

D.J. Evans, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 3 Use and calibration of vibration and shock measuring devices
13 Watch Hill Place, Gaithersburg, MD 20878

D.D. Reynolds, Chair, U.S. Technical Advisory Group for ISO/TC 108/SC 4 Human exposure to mechanical vibration and shock
3939 Briar Crest Court, Las Vegas, NV 89120

D.J. Vendittis, Chair of the U.S. Technical Advisory Group for ISO/TC 108/SC 5 Condition monitoring and diagnostics of machine systems
701 Northeast Harbour Terrace, Boca Raton, FL 33431

D. A. Preves and C. Walber, U.S. Technical Co-advisors for IEC/TC 29, Electroacoustics
(D. Preves) Starkey Hearing Technologies, 6600 Washington Ave., S., Eden Prairie, MN 55344
(C. Walber) PCB Piezotronics, Inc., 3425 Walden Ave., Depew, NY 14043 2495

The reports of the Chairs of these TAGs will not be presented at any other S Committee meeting.

The meeting of the Standards Committee Plenary Group will follow the meeting of Accredited Standards Committee S2, which will be held on Monday, 18 May 2015 from 5:00 p.m. - 6:00 p.m.

The Standards Committee Plenary Group meeting will precede the meetings of the Accredited Standards Committees S1, S3, S3/SC 1, and S12, which are scheduled to take place in the following sequence:

Tuesday, 19 May 2015	11:00 a.m. - 12:00 p.m.	S1, Acoustics
Tuesday, 19 May 2015	1:45 p.m. - 2:45 p.m.	ASC S3/SC 1, Animal Bioacoustics
Tuesday, 19 May 2015	3:00 p.m. - 4:15 p.m.	ASC S3, Bioacoustics
Tuesday, 19 May 2015	4:30 p.m. - 5:45 p.m.	ASC S12, Noise

Discussion at the Standards Committee Plenary Group meeting will consist of national items relevant to all S Committees and U.S. TAGs.

The U.S. Technical Advisory Group (TAG) Chairs for the various international Technical Committees and Subcommittees under ISO and IEC, which are parallel to S1, S2, S3 and S12 are as follows:

U.S. TAG Chair/Vice Chair	TC or SC	U.S. Parallel Committee
ISO		
P.D. Schomer, Chair	**ISO/TC 43** Acoustics	ASC S1 and ASC S3
P.D. Schomer, Chair	**ISO/TC 43/SCI** Noise	ASC S12
M.A. Bahtiarian, Chair	**ISO/TC 43/SC 3**, Underwater acoustics	ASC S1, ASC S3/SC 1 and ASCS12
W. Madigosky, Chair	**ISO/TC 108** Mechanical vibration, shock and condition monitoring	ASC S2
M. L'vov, Chair	**ISO/TC 108/SC2** Measurement and evaluation of mechanical vibration and shock as applied to machines, vehicles and structures	ASC S2
D.J. Evans, Chair	**ISO/TC 108/SC3** Use and calibration of vibration and shock measuring instruments	ASC S2
D.D. Reynolds, Chair	**ISO/TC 108/SC4** Human exposure to mechanical vibration and shock	ASC S3
D.J. Vendittis, Chair	**ISO/TC 108/SC5** Condition monitoring and diagnostics of machine systems	ASC S2
IEC		
D.A. Preves and C. Walber U.S. Technical Co-advisors	**IEC/TC 29** Electroacoustics	ASC S1 and ASC S3

TUESDAY MORNING, 19 MAY 2015 — DUQUESNE, 11:00 A.M. TO 12:00 P.M.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

R. J. Peppin, Chair ASC S1
5012 Macon Road, Rockville, MD 20852

A. A. Scharine, Vice Chair ASC S1
U.S. Army Research Laboratory, Human Research & Engineering Directorate
ATTN: RDRL-HRG, Building 459 Mulberry Point Road Aberdeen Proving Ground, MD 21005 5425

Accredited Standards Committee S1 on Acoustics. Working group chairs will report on the status of standards currently under development in the areas of physical acoustics, electroacoustics, sonics, ultrasonics, and underwater sound. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics, ISO/TC 43/SC 3, Underwater acoustics, and IEC/TC 29 Electroacoustics, take note-those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 19 May 2015.

Scope of S1: Standards, specifications, methods of measurement and test, and terminology in the field of physical acoustics, including architectural acoustics, electroacoustics, sonics and ultrasonics, and underwater sound, but excluding those aspects which pertain to biological safety, tolerance, and comfort.

Session 2pAA

Architectural Acoustics, Noise, Speech Communication, Psychological and Physiological Acoustics, and Signal Processing in Acoustics: Sixty-Fifth Anniversary of Noise and Health: Session in Honor of Karl Kryter II

Larry E. Humes, Cochair
Dept. of Speech & Hearing Sci., Indiana Univ., Bloomington, IN 47405-7002

Kenric D. Van Wyk, Cochair
Acoustics By Design, Inc., 124 Fulton Street East, Second Floor, Grand Rapids, MI 49503

Chair's Introduction—1:00

Invited Papers

1:05

2pAA1. The cardiovascular effects of noise on man. Wolfgang Babisch (Himbeersteig 37, Berlin 14129, Germany, wolfgang.babisch@t-online.de)

Noise is pervasive in everyday life and induces both auditory and non-auditory health effects. Systematic research of the effects of noise on the cardiovascular system has been carried out for more than 50 decades. Noise is a stressor that affects the autonomic nervous system and the endocrine system. Animal experiments, laboratory and field studies carried out on humans provide evidence that persistent exposure to environmental noise affects physiological endpoints, which in turn are adversely associated with cardiovascular diseases. These include hypertension, coronary heart disease, and stroke. New endpoints have been studied, including clinical states of metabolic syndrome such as diabetes mellitus. Chronic sleep disturbance is considered as an important mediator of the effects. Public health policies rely on quantitative risk assessment to set environmental quality standards and to regulate the noise exposure that is generated by environmental noise sources in the communities. Meta-analyses were carried out to derive exposure-response relationships between noise and cardiovascular health. Most of the epidemiological studies refer to road traffic and aircraft noise. No biologically determined threshold values can be determined. Cardiovascular effects due to noise and traffic-related air pollutants are largely independent of one another due to different biological mechanisms.

1:35

2pAA2. The behavioral impacts and spatial dynamics of alarm fatigue in reverberant healthcare treatment spaces. Paul Barach (Univ. of Oslo, 31 Baker Farm Rd., Lincoln, Massachusetts 01773, pbarach@gmail.com)

Medical device alarms are deliberately designed to alert attention. Conditions found in hospitals produce the unintended consequences triggered by alarms called "alarm fatigue" that is not attributable to individually alarmed devices but rather to the aggregate conditions in which the alarms occur. "Alarm fatigue" is a condition of recurrent noises from myriad uncorrelated medical devices, set at maximum loudness, occurring in hard-walled, reverberant spaces and produces elevated stress, sleep impairment, disorientation and dangerously irrational behavior. Four conditions cause "alarm fatigue": (1) burgeoning use of alarmed devices none of which are prioritized or correlated with others and all of which are set to maximum sound pressure levels; (2) advanced technologies that have caused a steady increase in hospital noise levels; (3) small enclosed spaces that surround dense clusters of healthcare workers and patients with hard, sound-reflecting surfaces and high levels of noise; and, (4) codes and standards that require alarms to be significantly louder than background, ambient sound. These conditions produce an "acoustic feedback loop" in which noise inevitably and rapidly escalates to intolerable levels and interferes with behavior. Understanding these conditions and developing practical and effective ways to interrupt "acoustic feedback loops" requires trans-disciplinary approaches and mixed implementation methods that include health service researchers, signal-processing, bioacoustics, and room acoustics.

1:55

2pAA3. Transdisciplinary, clinical analysis of the effects of noisy environment on the cognitive performance of healthcare providers. Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180, xiangn@rpi.edu), David Sykes (Acoust. Res. Council, Boston, MA), Wayne R. Triner (Emergency Medicine, Albany Medical College, Albany, NY), and Peter Dodds (School of Architecture, Rensselaer Polytechnic Inst., Troy, NY)

Karl Kryter's pioneering and encyclopedic effort begun over six decades ago to document "The Effects of Noise on Man" opened a door to trans-disciplinary, translational research. His approach continues to challenge many of us in acoustical science, medicine, and the development of codes and standards. The authors have emulated Kryter's trans-disciplinary approach by forming a unique and stable

partnership between doctors at a major medical center, scientist-engineers at a polytechnic institute, and a division of the American Hospital Association that promulgates evidence-based building criteria. This partnership enables detailed and insightful, quantitative research to be conducted in a variety of previously inaccessible clinical environments (e.g., hospitals) on the impacts of ambient noise on the cognitive performance of healthcare professionals. The authors couple psychoacoustic methods and soundscape analysis with cognitive performance tests to examine questions of major concern in the healthcare professions such as the extent to which medical errors are caused by environmental stresses such as noise. Several studies are currently being designed that highlight the challenges and opportunities of emulating Kryter's approach using 21st century methods.

2:15

2pAA4. Assessing hazardous noise pollution in the United States. Richard L. Neitzel (Dept. of Environ. Health Sci. and Risk Sci. Ctr., Univ. of Michigan, Ann Arbor, MI), Monica S. Hammer, and Tracy K. Swinburn (Risk Sci. Ctr., Univ. of Michigan, 1415 Washington Heights, Ann Arbor, MI 48109, monicashammer@gmail.com)

The risk of noise-induced hearing loss (NIHL) from noise exposure has been known for hundreds of years, but Karl Kryter's seminal work on NIHL represented a major step forward in addressing this disease. In recent years, a growing body of evidence has linked noise to additional non-auditory health effects, including cardiovascular disease, sleep disturbance, and stress. The United States has no national plan to address noise pollution, although standards designed to protect human health from noise do exist. National estimates of U.S. exposures were last created in the early 1980s. We have updated these estimates using current census and research data, and estimate that 104 million individuals in the United States had annual equivalent continuous average levels >70 dBA in 2013 and were at risk of NIHL and other effects. We describe approaches to the assessment of noise exposures at the level of the individual and the community, and outline ways in which noise can be more comprehensively assessed. Based on the number of exposed individuals in the United States and the impact of noise-related health effects, greater emphasis on noise reduction will likely improve public health and quality of life. This increased focus will allow us to continue and expand upon Kryter's outstanding work.

2:35

2pAA5. Karl Kryter—Soundscape pioneer. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooksacoustics.com)

Karl Kryter was a pioneer in the soundscape analysis field, even before the field had a name. Defined as the combination of the study of physical sound parameters with the perception of sound by listeners in an environmental context, soundscape analyses were conducted and reported by Kryter well before the term "Soundscape" was coined by R. Murray Schafer in 1977. These included several studies of community reactions to noise from subsonic aircraft while Kryter was with Bolt, Beranek and Newman in the 1950's. Those studies were followed by laboratory and field tests of listener reactions to aircraft noise in the 1960's. Kryter compiled and compared similar attitudinal and perceptional studies of transportation noise by other researchers in his 1970 book "The Effects of Noise on Man". This compilation has proven to be comprehensive and enduring in its presentation of noise perception data. Kryter's insights from those early studies provide valuable guidance to current and future soundscape researchers and soundscape design practitioners.

Contributed Papers

2:55

2pAA6. Contribution of floor treatment characteristics to background noise levels in health care facilities, Part 2. Adam C. Ackerman, Jonathan Zonenshine, Eoin A. King, Robert D. Celmer (Acoust. Prog. & Lab, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, celmer@hartford.edu), and John J. LoVerde (Paul S. Veneklasen Res. Foundation, Santa Monica, CA)

Additional acoustical tests were conducted as Part 2 of a previous study [JASA **136**(4), 2219–(2014)] on ten types of commercial grade flooring to assess their potential contribution to surface-generated noise within healthcare facilities. These floor samples utilized an ECORE *ForestFX* sheet vinyl wear layer and tested with rubber-backing thicknesses ranging from 2 mm–10 mm, plus a composite backing composed of two 2 mm layers. Two types of ECORE International adhesives, *EGripIII* and *Evolve*, were tested as part of the adhesive used during manufacturing as well as with local mounting of all floors on concrete substrates. Sound power tests were conducted in compliance with ISO-3741-2010 for two source types, an impact tapping machine and a rolling hospital cart. The sample that achieved the lowest radiated sound power levels for both the rolling cart and tap tests was *Forest FX* sheet vinyl with 10 mm rubber backing and Evolve adhesive. Two trends were observed. First, measured sound power levels reduced as backing thickness rose. Second, the *Evolve* adhesive demonstrated lower sound power levels for all tap/rolling tests when compared with those utilizing the *EGripIII* adhesive. The suitability of using thicker floor backings in healthcare facilities are discussed. [Work supported by Paul S. Veneklasen Research Foundation.]

3:10

2pAA7. Effect of adhesives in structure-borne and impact sound measurements. Sharon Paley (ECORE Int.., 715 Fountain Ave., Lancaster, PA 17601, sharon.paley@ecoreintl.com) and Bradlay Hunt (ATI/Intertek, York, PA)

Based on a preliminary finding in "Contribution of Floor Treatment Characteristics to Noise Levels in Health Care Facilities Part 2," we investigate the effect that various adhesives may have in surface generated sound and impact sound transmission levels. The aforementioned study demonstrated that lower source room sound levels were generated with an acrylic adhesive assembly when compared to sound levels generated with a urethane adhesive assembly. These findings led to two follow-up questions: (1) What effect does adhesive have in IIC ratings? and (2) Is there a clear inverse relationship between sound levels within a source and receiving room, and how much does the adhesive affect this relationship? Results from tests conducted in a lab environment with multiple floor finishes and multiple assemblies will be discussed.

3:25–3:35 Break

Invited Papers

3:35

2pAA8. Karl Kryter: The evolution of the Articulation Index. Jont B. Allen (ECE, Univ. of IL, 1404 Sunny Acres Rd., Mahomet, IL 61853, jontallen@ieee.org)

When one trained in speech perception hears the name Karl Kryter, they immediately think of the AI standard, which Kryter ferried through the standards process. Today, the AI has been studied to the point that we are beginning to understand both its beauty, and its fundamental limitations. The AI was created by Fletcher and Galt, as reported first in JASA in 1950, and in Fletcher's two books. It is based on two important intertwined observations: (1) that the average consonant error Pe is exponential in an average of critical band signal to noise ratios (i.e., $P_e(SNR) = 0.015{AI}$) and (2) the total error is the product of independent critical band errors. While AI theory has been verified many times over, the real question is "Why does it work?" In this talk, I will answer this question, and in the process expose the AIs key weaknesses. AI theory has guided us to our modern understanding of speech perception in normal and hearing impaired ears. The publications supporting this work date from 1994–2014 and may be found at http://hear.ai.uiuc.edu/Publications.

3:55

2pAA9. The use of articulation theory and the speech intelligibility index in the design of clinical speech perception tests. Douglas Brungart and Kenneth W. Grant (Walter Reed NMMC, 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.brungart@us.army.mil)

For more than 50 years, the Articulation Index (AI) and its successor the Speech Intelligibility Index (SII) have been recognized as essential tools for predicting the impact that reduced bandwidth, background noise, and differences in speech materials will have on the speech perception performance of normal hearing listeners. However, one caveat about both AI and SII is that neither is considered to be valid for predicting the performance for hearing impaired listeners. Nevertheless, the insights provided by the AI and SII can be invaluable in the design and analysis of clinical speech tests designed to evaluate the relative performance of hearing impaired listeners. In this talk, we discuss role that the SII and AI played in the development of a speech-in-noise test based on the Modified Rhyme Test that is intended for use in the evaluation of Auditory Fitness-for-Duty in military personnel. In this case, the information drawn from the SII was essential both in the selection of appropriate speech materials and SNR values and in finding ways to analyze the results of normal and hearing impaired listeners. The results of the study provide evidence for a "pure" SNR loss in hearing impaired listeners that occurs even when all components of the speech signal are audible. [The views expressed in this article are those of the authors and do not necessarily reflect the official policy or position of the Department of Defense or U.S. Government.]

4:15

2pAA10. A challenge to Articulation Theory: Narrow-band acoustic signals and visual speech cues. Ken W. Grant (Audiol. and Speech Ctr., Walter Reed National Military Med Ctr., 301 Hamilton Ave., Silver Spring, MD 20901, ken.w.grant@gmail.com)

Given Kryter's extensive work on the physiological and subjective effects of noise on the human auditory system it is not surprising that at some point he would turn his attention to the impact of noise on speech intelligibility. In 1962, Kryter published "Methods for the Calculation and Use of the Articulation Index" greatly simplifying the methods originally proposed by Fletcher in the 20s and 30s and French and Steinberg in 1947, ultimately leading to the ANSI 1969 standard. Over the years, the AI and the revised Speech Intelligibility Index (SII) have become the leading metric for predicting speech recognition in noise and in quantifying the audibility provided by amplification. However, the scope of the AI/SII is still limited to communication conditions, which do not include multiple, narrow bands of speech or noise. Similarly, predictions for conditions of auditory-visual speech are only roughly approximated. Data from a variety of studies suggest that rather than ignore these communication conditions they should be studied closely for what they reveal. In this talk, I will show data that expose chinks in the armor of our understanding of speech perception in noise and under conditions where visual speech cues are present.

4:35

2pAA11. Predicting speech intelligibility for hearing-impaired listeners. James M. Kates (Speech Lang. Hearing Sci., Univ. of Colorado, 409 UCB, Boulder, CO 80309, James.Kates@colorado.edu)

Karl Kryter was instrumental in developing the Articulation Index (AI) standard and demonstrating its effectiveness. The AI, and its successor, the Speech Intelligibility Index (SII), predict intelligibility based on the audibility of speech sounds. The concept of audibility is also relevant to hearing loss and the effectiveness of hearing aids, and extensions of the AI have been developed to predict hearing-aid benefit. However, audibility alone cannot explain the impact of nonlinear distortion and nonlinear hearing-aid processing on intelligibility. This talk will present a review of the how the AI and SII have been used for hearing aids, extensions to the SII targeted at predicting hearing-aid effectiveness, and more recent work that focuses on envelope fidelity rather than audibility in predicting intelligibility for hearing-aid users.

4:55

2pAA12. The effects of noise on learners of English as a second language. Catherine L. Rogers (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu)

In the 65 years since the publication of The Effects of Noise on Man, as the world has grown ever noisier, it has also grown more diverse and more connected than ever before. Today, nearly a fifth of U.S. residents speak a language other than English at home. Consequently, exploring the effects of noise on people communicating in a second language is a crucial component of understanding the effects of noise on mankind in our increasingly pluralistic society. This presentation will overview some of the challenges faced by second-language learners as speakers and listeners, particularly focusing on the extraction of phonetic information from a noisy or

impoverished signal. Results of ongoing research investigating vowel identification by native and non-native English-speaking listeners will be presented. Overall accuracy, the slope of the function relating signal-to-noise ratio and intelligibility, and effects of noise on listeners' ability to benefit from phonetic enhancement strategies will be compared across native English speakers and second-language learners with differing ages of immersion in an English-speaking environment.

5:15–5:30 Panel Discussion and Open Microphone

TUESDAY AFTERNOON, 19 MAY 2015 RIVERS, 1:00 P.M. TO 5:40 P.M.

Session 2pAO

Acoustical Oceanography and Underwater Acoustics: Acoustics of Fine Grained Sediments: Theory and Measurements

Mohsen Badiey, Cochair
College of Earth, Ocean, and Environment, University of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716

David P. Knobles, Cochair
Univ. of Texas, Austin, PO Box 8029, Austin, TX 78713

Charles W. Holland, Cochair
Applied Research Laboratory, The Pennsylvania State University, P.O. Box 30, State College, PA 16804

Chair's Introduction—1:00

Invited Papers

1:05

2pAO1. Modeling marine mud: A four phase system. Richard H. Bennett (SEAPROBE, Inc., 501 Pine St., Picayune, MS 39466, rhbenn_seaprobe1@bellsouth.net) and Matthew H. Hulbert (Res. Dynam., West Chester, PA)

Fine-grained marine sediment consists of up to four phases: (1) clay Minerals often with some silt and sand, (2) saline water, (3) free gas, and (4) organic matter (OM). Shallow-water, shelf, and slope muds often have all four phases present. The clay minerals are the "building blocks" of mud deposits; these consist of multi-plate face-to-face domains. There are only three possible modes of domain contact (termed signatures): (1) edge-to-face (EF), (2) edge-to-edge (EE), and (3) face-to-face (FF), usually offset. The domain faces are negatively charged and have a large surface area and the edges are both negative and positive charged. OM is usually expressed as total organic carbon (TOC) in percent total dry mass and can range from <0.5% to several percent and the OM can occupy a significant portion of pore volume. OM is usually attached to clay domain signatures and faces with the location partially dictated by electrostatic potential energy fields. When several percent TOC is present, the OM drives up water contents and porosity considerably. When free gas is present, it replaces part of the seawater in the pore space. Several published organo-clay fabric models for mud deposits may improve prediction of acoustic properties.

1:25

2pAO2. Frequency power-law attenuation and dispersion in marine sediments. Michael J. Buckingham (SIO, UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0238, mbuckingham@ucsd.edu)

Acoustic waves in marine sediments, including the fine-grained muds and clays, often exhibit an attenuation in the form of a frequency power law in which the exponent is close to unity. The frequency dependence of the wave speed, or dispersion, associated with such a power law has long been a subject of debate. Causality arguments, often in the form of the Kramers-Kronig dispersion relations, are usually applied to the problem, but these are not sufficient to characterize the dispersion fully. Following an alternative approach, a wave equation will be introduced which predicts the dispersion when the attenuation is of the frequency power law type. Based on this wave equation, it will be shown that only certain values of the frequency exponent are allowed, otherwise causality is violated, indicating that the associated attenuation and dispersion are not physically realizable. In effect, for these prohibited values of the frequency exponent, the Fourier components in the dispersion and attenuation cannot combine to give a zero response for negative times. Emerging from the theory is an expression for the complex wavenumber that is complete and exact, and from which all the properties of the dispersion and attenuation may be derived.

1:45

2pAO3. Geoacoustic modeling for acoustic propagation in mud ocean-bottom sediments. William L. Siegmann (Dept. of Mathematical Sci., Rensselaer Polytechnic Inst., 110 Eighth St., Troy, NY 12180-3590, siegmw@rpi.edu) and Allan D. Pierce (Retired, East Sandwich, MA)

This paper reviews research developed since 2007 at BU and RPI, with leadership by the late W.M. Carey, on acoustic properties of mud. Marine mud consists primarily of small clay mineral platelets, which are comprised of crystalline layers and usually carry charge because of isomorphous substitution. Because of resulting electrical forces, the physical nature of mud is considerably different from sand sediments. In particular, as platelets settle under gravity, electrical forces repel face-to-face contact while strong van der Waals forces permit edge-to-face attachment. This platelet aggregation results in card-house structures, for which a tentative quantitative model has been analyzed (J.O. Fayton, RPI Ph.D. thesis). The model preserves basic physics of platelet interactions and leads to low shear wave speed predictions that are consistent with observations. However, because compressional sound speed is independent of the electrostatic interactions, accurate sound speed estimates are available from the Mallock-Wood formula, which also incorporates bubble effects. The basic physical concepts and semi-empirical formulas derived for shear attenuation and its frequency dependence in sandy sediments do not apply for mud, nor do assumptions behind the often cited Biot theory for poro-elastic media. Consideration is given to incorporating geoacoustic models into equations for acoustic propagation in mud.

2:05

2pAO4. *In situ* measurements of sediment compressional and shear wave properties in Currituck Sound. Megan S. Ballard, Kevin M. Lee, Andrew R. McNeese, Preston S. Wilson, Thomas G. Muir (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu), and R D. Costley (U.S. Army Engineer Res. and Development Ctr., Vicksburg, MS)

This paper reports *in situ* measurements of compressional and shear wave speed and attenuation collected below the sediment-water interface in Currituck Sound, North Carolina. Three measurement locations having distinctly different sediment types will be presented: (1) a near shore site with coarse sand, (2) a shallow-water site with silty fine-grained sand, and (3) an offshore site with loamy clay. At each site, grab samples were collected and later analyzed in the laboratory to quantify physical properties of the sediment, including grain size, density, and porosity. The *in situ* acoustic data were acquired using two measurement systems, each consisting of four probes mounted on a rigid frame. The shear wave system consisted of bimorph elements to generate and receive horizontally polarized shear waves; the compressional wave system was composed of cylindrically shaped piezoelectric elements. Both systems were manually inserted into the sediment, and waveforms were recorded by shipboard equipment. The measured wave speeds and attentions are compared to predicted values from one or more sediment models using the measured physical properties as inputs. The Biot-Stoll model, grain-shearing theory, Mallock-Wood equation, and card-house theory are considered for this analysis. [Work supported by ERDC, ARL:UT, and ONR.]

Contributed Papers

2:25

2pAO5. Tentative acoustic wave equation that includes attenuation processes for mud sediments in the ocean. Allan D. Pierce (Retired, PO Box 339, East Sandwich, MA 02537, allanpierce@verizon.net) and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

Compressional wave attenuation in mud sediments is less than in sandy/silty sediments and comparable to that of sea water. Experimental results by Wood and Weston (Acustica,1964) yield 0.07 dB/m for a frequency of 1 kHz, while the extensive data analysis of Holmes, Carey, Dediu, and Siegmann (JASA-EL, 2007) suggests a value of 0.33 dB/m for sandy/silty sediments. The linear frequency dependence over the range of 4 to 50 kHz reported by Wood and Weston suggests that the cause of the attenuation is relaxation processes rather than viscosity processes. The tentative appropriate relaxation process is suggested, in accord with previous ideas reported by Mahmood et al. (Colloids and Surfaces A, 2001), to be the breaking and reforming of van der Waals-force-related bonds between clay platelets that are connected edge to face. The present paper's theoretical development of a single wave equation with relaxation included parallels that discussed on pp. 550–553 of the text by Pierce (Acoustics, 1981) and follows the ideas of Liebermann (Phys. Rev., 1949). Predicted sound speed is in accord with the Mallock-Wood equation, and the attenuation term appears as an integral over past history, in accord with the envisioned frequency-dependent attenuation.

2:40

2pAO6. Fine-grained sediments from the poroelastic perspective. Nicholas P. Chotiros and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas at Austin, PO Box 8029, Austin, TX 8713-8029, chotiros@arlut.utexas.edu)

Fine-grained sediments, such silt and clay, are porous media, consisting of solid particles saturated by water; however, the skeletal structure of the solid component is significantly different than that of larger grained sediments such as coarse and fine sands. The modeling of fine grained sediments from the poroelastic perspective, such as the Biot theory, can be challenging. A stumbling block is the frequency dependence of attenuation, which from experimental measurement is shown to be proportional to the first power of frequency, while the Biot theory generally predicts a second power of frequency dependence at low frequencies. One approach to reconcile this difference is based on the distribution of pore sizes [Yamamoto and Turgut, 1988]. When a broad distribution of pore sizes in taken into consideration, the frequency dependence approaches the first power of frequency within a limited frequency band. Another approach is to examine the relaxation process at the grain-grain contact, which is governed by a thin fluid film in the contact region. In fine-grained sediments, the contact region is extremely small and nano-fluid effects must be expected. In this study, a derivation of a poroelastic model, based on the relaxation process associated with the grain contact region will be discussed. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

2:55–3:10 Break

3:10

2pAO7. Modeling transmission loss over stratified muddy sediments based on chirp sonar imagery. Dajun Tang, Brian T. Hefner, Jie Yang, and Darrell R. Jackson (Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

The Office of Naval Research is sponsoring a shallow water acoustics experiment offshore New Jersey where the sediment consists nominally of a layer of mud over sand, and where the layer thickness is about 10 m. A preliminary chirp sonar survey shows that there exists fine layering within the mud layer. Nearby core data also reveal the existence of fine layers within the mud. One of the interesting acoustic questions is what impact such fine layering within the muddy sediments has on acoustic propagation. Based on the chirp sonar imagery, various plausible models of sediment layering and material composition are proposed and the transmission loss is estimated for those models for the purpose of assisting field experiment planning.

3:25

2pAO8. High-frequency scattering from a sand sediment with an overlying mud layer. Brian T. Hefner, Anatoliy Ivakin, and Darrell R. Jackson (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, hefner@apl.washington.edu)

In the spring of 2014, multibeam echo sounder time series data were collected in St. Andrew's Bay, FL, in an area of the bay where the sand sediment was covered by a mud layer. As part of the environmental characterization at the experiment site, the *In-Situ* Measurement of Porosity (IMP2) system collected conductivity probe data 25 cm into the seabed along 3 m tracks. The mud layer appears clearly in the conductivity probe data and had a mean thickness of 13 cm. The roughness power spectrum of the sand/mud interface was 10–20 dB higher than that of the mud/water interface and, more significantly, was 5–10 dB higher than that of sand/water interfaces measured in the Gulf of Mexico during the Target and Reverberation Experiment 2013. The mud layer appears to be preserving the roughness of the sand interface, an effect observed during the Sediment Acoustics Experiment in 2004 following the passage of Hurricane Ivan. The impact of both the increased roughness and the presence of the mud on 180–420 kHz scattering will be assessed through data/model comparisons using the sediment properties measured at the experiment site. [Work supported by ONR and SERDP.]

3:40

2pAO9. Seabed attenuation inversion from broadband reverberation measurements in the Yellow Sea. Lin Wan and Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., 104 Robinson Hall, Newark, DE 19716, wan@udel.edu)

The Yellow Sea 1996 experiment was the first joint China-U.S. shallow water experiment conducted in August, 1996. The water depth is 75 m with a deviation of ± 1 m at the experimental area. The seabed sound speed and attenuation in this experiment have been successfully estimated using the sound propagation measurements [Wan, Zhou, and Rogers, J. Acoust. Soc. Am. **128**(2), 2010]. During this experiment, the monostatic reverberation data produced by 1-kg TNT shots provide a spatial sampling of the acoustic field and contains the information of seabed acoustic properties. Thus, the present paper uses the inverted seabed sound speed from the aforementioned propagation measurements as a constraint condition and analyzes the reverberation data for seabed attenuation inversion. The broadband normal mode filter is applied to the reverberation data received by a 32-element vertical line array. The reverberation level of the first three modes is obtained as a function of range to extract the modal attenuation coefficients. The seabed attenuation is inferred by minimizing the difference between the theoretical and measured modal attenuation coefficients. The inverted seabed attenuation over a frequency range of 400–1000 Hz shows non-linear frequency dependence.

3:55

2pAO10. The angle of intromission in muddy sediments: Measurements and inferences. Charles W. Holland (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, cwh10@psu.edu), Jan Dettmer, and Stan E. Dosso (School of Earth and Ocean Sci., Univ. of Victoria, Victoria, BC, Canada)

Fine-grained sediments, e.g., mud, often exhibit sound speeds less than that of the interstitial fluid. This leads to an angle of intromission (AI), or Brewster angle, at which there is nearly complete transmission of acoustic energy from the water column into the sediment. It is generally difficult to observe this angle directly inasmuch as (1) fine-grained sediments typically have a very low attenuation and (2) underlying granular sediments have a higher sound speed and thus the total seabed reflected field tends to be dominated by the underlying more highly reflective sediments. In some cases, the AI can be directly observed by time-gating the reflected field such that only the fine-grained sediments are insonified. Nominally, the AI is independent of frequency. However, reflection measurements from hundreds of hertz to several kilohertz show a frequency dependence. This dependence is due to sound speed and density gradients (which thus can be estimated from the reflection data). Reflection measurements and core data indicate that geoacoustic gradients depend upon location on the continental shelf. In the Italian littoral, density gradients are always observed to be positive and decrease seaward. Curiously, sound speed gradients can be positive or negative and have a less obvious correlation with position. [Research supported by ONR Ocean Acoustics.]

4:10

2pAO11. A semi-empirical predictive model for the attenuation properties of fine-grained sediments. David P. Knobles, Steven A. Stotts, and Robert A. Koch (ARL, Univ. of Texas, PO Box 8029, Austin, TX 78713, knobles@arlut.utexas.edu)

It is of interest to be able to predict the acoustic properties (especially the compressional sound speed and attenuation) of sediments that can be classified as mud. An inverse problem is constructed to infer estimates of the surface sediment attenuation, the attenuation gradient, and the frequency exponent and applied to acoustic data in the 10–400 Hz band collected in the Gulf of Oman basin. The results indicate that, unlike a sandy seabed, the surface attenuation is about 0.005 dB/m-kHz and the frequency exponent is close to unity. These estimates are compared to those of an inversion analyses for a shallow-water site in the Gulf of Mexico, which is an environment with similar sediment properties. Further, the inferred attenuation values from both experiments are compared to calculated values from the Buckingham grain-shearing model. The geological histories of the shallow- and deep-water seabeds are compared with the intent of constructing a semi-empirical predictive model for the attenuation properties of fine-grained sediments.

4:25

2pAO12. Shear wave attenuation estimates from Scholte wave data. Gopu R. Potty, James H. Miller (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu), and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., Univ. of California, San Diego, CA)

Scholte waves are valuable tool for estimating the shear properties of the ocean bottom sediments. Previously estimates of the shear wave speed were obtained using interface wave data from a small scale experiment conducted in very shallow water in coastal Rhode Island. The University of Rhode Island's shear measurement system consisting of vertical axis and three-axis geophones were used to collect data in 3 m of water. Interface waves were excited by dropping a weight from a research vessel. In this study, we use the geophone data to estimate the depth-averaged shear attenuation using spectral amplitude ratios. The estimated sediment properties will be compared with historic core data from the field test location and other published results for similar type of sediments. The correlation of the shear speed and attenuation to sediment properties such as porosity, grain size and bulk density will be discussed. [Work supported by Office of Naval Research.]

4:40

2pAO13. Wave propagation in muddy sediments using time domain finite difference approach. Holly C. Clark (R&D Ocean Eng., Appl. Physical Sci., 475 Bridge St., Groton, CT 06340, hclark@aphysci.com), Ralph A. Stephen (Geology and Geophys., Woods Hole Oceanographic Inst., Woods Hole, MA), James H. Miller, and Gopu R. Potty (Ocean Eng., Univ. of Rhode Island, Narragansett, RI)

There is a renewed interest in understanding wave propagation in muddy sediments. In this study, we are using the time domain finite difference approach to investigate wave propagation in a muddy environment representative of the New England Mud Patch south of Martha's Vineyard. The model uses the two-dimensional full-wave time-domain finite-difference code developed at WHOI over the past 35 years. In order to simplify the computation we consider perfectly elastic, isotropic media with uniform step sizes in time and space. The source waveform is a Ricker wavelet with a peak frequency in pressure of 200 Hz. This study considers a 100 m water waveguide (Vp = 1500 m/s) overlying 12 m of mud (Vp = 1530 m/s, Vs = 100 m/s) above a faster bottom (Vp = 1700 m/s, Vs = 500 m/s). The goal of the modeling was to understand the propagation of compressional, shear and interface waves in the presence of a soft muddy layer. [Work supported by ONR Code 3220A.]

4:55

2pAO14. Efficient geoacoustic inversion of spherical-wave reflection coefficients for muddy seabeds. Jorge E. Quijano, Stan E. Dosso, Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr. A405, 3800 Finnerty Rd. (Ring Road), Victoria, British Columbia V8P 5C2, Canada, jorgeq@uvic.ca), and Charles W. Holland (ARL, Penn State Univ., Reston, VA)

This paper presents efficient geoacoustic inversion of high-resolution reflectivity data for a seabed site consisting of a thick mud layer (~10 m) over multi-layered sandy sediments. Wide-angle, broadband reflection-coefficient data were collected using a towed source and a recording hydrophone located a few meters above the seabed. Trans-dimensional Bayesian inversion is applied to estimate sound-speed, density, and attenuation profiles for an unknown number of sediment layers. Due to the experiment geometry, the data should be modeled as spherical-wave reflection coefficients, which require numerical integration of rapidly oscillating functions and can be computationally intensive. Here, a speed-up of two to three orders of magnitude is achieved by introducing a fast Levin-type numerical integration implemented on a graphics processing unit. The new fast integration algorithm/implementation alleviates time constraints for the spherical-wave inversion, which would otherwise require weeks of computation time, and precludes the use of fast but less-accurate plane-wave theory. Inversion results are presented for simulated data and for experimental data collected on the western Malta Plateau. The analysis of muddy sediments in this work is expected to provide insight into the geoacoustics of the New England mud patch, location of the upcoming ONR Seabed Characterization Experiment 2016 (SCAE16).

5:10

2pAO15. Measurement of shear wave velocity in marine sediments. Aaron S. Bradshaw (Civil and Environ. Eng., Univ. of Rhode Island, 207A Bliss Hall, Kingston, RI 02818, bradshaw@egr.uri.edu) and Christopher D. Baxter (Ocean Eng. and Civil Eng., Univ. of Rhode Island, Narragansett, RI)

This presentation describes the shear wave velocity measurement techniques that have been utilized over that past decade at the Marine Geomechanics Laboratory (MGL) at the University of Rhode Island (URI). The shear wave velocity of marine sediments is of interest to both the underwater acoustics and marine geotechnical engineering communities. In underwater acoustics, sediment shear wave velocity plays an important role in understanding compressional wave attenuation. Geotechnical engineers utilize shear wave velocity, for example, for the analysis of seismic site response and evaluation of soil liquefaction potential. The shear wave measurement techniques that will be covered include laboratory bender element tests made within the triaxial apparatus that allows for the application of situ stress states on the test specimen. Bender elements have also been used in the lab to make shear wave velocity measurements in reconstituted sand fills at very low stress levels, as well as on intact clay sediments directly within a piston core. More recently, field measurements of shear wave velocity have been made using a system developed by colleagues at URI that involves the measurement and inversion of Scholte waves. The presentation aims to stimulate dialog between the fields of underwater acoustics and marine geomechanics on the topic of shear wave velocity.

5:25

2pAO16. Laboratory and field measurements of Rayleigh waves. Christopher J. Norton (Dept. of Civil Eng., Univ. of Rhode Island, 215 South Ferry Rd., Narragansett, RI 02882, cnorton@my.uri.edu), James H. Miller, Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Aaron Bradshaw, and Christopher D. Baxter (Dept. of Civil Eng., Univ. of Rhode Island, Kingston, RI)

There is a need for rapid and nondestructive sensing of near-surface shear properties of the ground and seafloor. Our approach on land is to measure Rayleigh wave dispersion and invert these measurements to extract a shear wave speed profile. A portable surface wave inversion system has been developed that employs an array of six accelerometers to measure Rayleigh waves excited by simple impulsive sources in the laboratory and in the field. To verify the shear wave speed estimated by the Rayleigh waves, a calibrated bender element system was also developed to provide independent measurements of the shear wave speed profile at selected depths. The bender element system was placed in a test tank initially filled with mason sand and the Rayleigh wave inversion system was deployed on the surface of the sand in the tank. The estimated shear wave speeds were then compared to the results of the bender element system as well measurements by a dynamic cone penetrometer and laser vibrometer. The approach allows for the measurement of shear speeds in for various sediment and soil arrangements in the well-controlled laboratory setting as well in the field. [Work supported by Army Research Office and ONR.]

Session 2pBA

Biomedical Acoustics: Ultrasonic Characterization of Bone

James G. Miller, Chair
Physics, Washington U Saint Louis, Box 1105, 1 Brookings Drive, Saint Louis, MO 63130

Invited Papers

1:00

2pBA1. Application of ultrasound transit time spectroscopy to human cancellous bone for derivation of bone volume fraction *in-vitro*. Christian M. Langton and Marie-Luise Wille (Inst. of Health & Biomedical Innovation, Queensland Univ. of Technol., 60 Musk Ave., Brisbane, Queensland 4049, Australia, christian.langton@qut.edu.au)

We have previously demonstrated that ultrasound propagation in complex composite media may be described as an array of parallel sonic rays. The transit time of each sonic ray is determined by the proportion of solid (bone) and fluid (marrow) traversed, the received ultrasound signal being a superposition of all sonic rays. An Ultrasound Transit Time Spectrum (UTTS) for a test sample may be obtained via digital deconvolution of input and output ultrasound signals, describing the proportion of sonic rays having a particular transit time, from which the bone volume fraction (BVF) of the sample may be estimated. In a recent *in-vitro* study, 21 cancellous bone samples, extracted from 5 human femoral heads following total hip replacement, were measured with microCT to derive the true BVF value. Transmission ultrasound signals of 1 MHz were recorded and UTTS-derived BVF calculated. A coefficient of determination (R^2) of 82% was achieved between ultrasound and microCT derived BVF values. Current work is clinically implementing UTTS, noting its potential to estimate bone mineral density, and hence, a means to diagnose osteopenia and osteoporosis using WHO T-score criteria.

1:20

2pBA2. A review of basic to clinical studies of quantitative ultrasound of cortical bone. Pascal Laugier, Simon Bernard, Quentin Vallet, Jean-Gabriel Minonzio, and Quentin Grimal (Laboratoire d'Imagerie Biomedicale, Universite Pierre et Marie Curie-Paris6, CNRS, INSERM, 15 rue de l'ecole de medecine, Paris 75017, France, laugierp@gmail.fr)

The use of quantitative ultrasound (QUS) for bone has increased sharply over the last two decades. QUS offers potential benefits over other diagnostic modalities which include (1) *ex vivo* non-destructive testing, (2) *in vivo* non-ionizing testing, and (3) the ability to assess important cortical bone quality factors which cannot easily be captured with X-ray techniques. These advantages have stimulated widespread interest in basic through clinical studies. For instance, resonant ultrasonic spectroscopy (RUS) has provided gains in *ex vivo* assessment of the anisotropic stiffness and viscoelasticity of bone despite strong damping. RUS is prone to provide answers to questions that remain open regarding the determinants of cortical bone elastic properties. *In vivo* QUS technologies using guided waves (GW) have the potential to reveal cortical bone strength-related factors such as the cortical thickness and porosity. These properties can be estimated by comparing the measured dispersion curves with an appropriate waveguide model of the cortical bone. This presentation reviews the *ex vivo* (RUS) and clinical (GW) studies of cortical bone based on major experimental studies particularly within the past decade. These gains still constitute a prelude to what is to come, given the incessant developments of better instrumentation and signal processing techniques.

1:40

2pBA3. A novel ultrasound device for estimating bone mineral density at the 1/3 radius. Jonathan J. Kaufman (CyberLogic, Inc. & The Mount Sinai School of Medicine, 611 Broadway Ste. 707, New York, NY 10012, jjkaufman@cyberlogic.org), Emily Stein (Columbia Univ. College of Physicians and Surgeons, New York, NY), Gangming Luo (CyberLogic, Inc., New York, NY), Elizabeth Shane (Columbia Univ. College of Physicians and Surgeons, New York, NY), and Robert S. Siffert (OrthopeDC, The Mount Sinai School of Medicine, New York, NY)

This study evaluated a new ultrasound device for estimating bone mineral density (BMD) at the 1/3 radius (1/3R), as well as investigated its ability to discriminate fracture (fx) and non-fx cases. The device measures two net time delay parameters, NTD_{DW} and NTD_{CW}. NTD_{DW} is the difference between transit time of an ultrasound pulse through soft-tissue, cortex and medullary cavity, and transit time through soft tissue of equal distance. NTD_{CW} is the difference between transit time of an ultrasound pulse through soft-tissue and cortex only, and transit time through soft tissue of equal distance. The square root of the product of these two parameters is a measure of BMD at the 1/3R as measured by DXA. A clinical IRB-approved study measured 77 adults using ultrasound and DXA. An age and sex-matched subset of these subjects was used to determine the capability of DXA and ultrasound to discriminate between fx and non-fx cases. A linear regression showed that $BMD_{US} = 0.19 * \{NTD_{DW} * NTD_{CW}\}^{1/2} + 0.28$ and that the linear correlation between BMD_{US} and BMD_{DXA} was 0.93 ($P<0.001$). We found that ultrasound measurements yield results that are closely associated with those from DXA. In the case-control fx study, we found a small decrease in mean BMD_{DXA} (1/3R) for the fx cases ($P=0.20$) but a somewhat greater decrease in mean BMD_{US} ($P=0.05$). In conclusion, the new ultrasound device should enable significant expansion of the identification of bone loss as occurs in osteoporosis and also may ultimately be able to provide additional data on fx risk.

2:00

2pBA4. Piezoelectric response of bone in the MHz range. Mami Matsukawa, Sayaka Matsukawa, and Hiroko Tsuneda (Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan, mmatsuka@mail.doshisha.ac.jp)

The healing mechanism of bone fractures by low intensity ultrasound is not yet to be fully understood. As one possible initial process of this mechanism, we focus on the piezoelectricity of bone and demonstrate that bone can generate electrical potentials by ultrasound irradiation in the MHz range. We have fabricated ultrasonic bone transducers using bovine cortical bone as the piezoelectric device. Electrical potentials induced by the ultrasound irradiation were obtained from all transducers. The electrical potentials changed as a function of time during immersed ultrasonic measurements and became stable when the bone was fully wet. The sensitivities of bone transducers were around 1/1000 of a poly(vinylidene fluoride) ultrasonic transducer and did not depend on the magnitude and alignment of hydroxyapatite crystallites in bone. In addition, the magnitude of the induced electrical potentials changed owing to the microstructure in the cortical bone. The potentials of transducers with haversian structure bone were higher than those of plexiform structure bone, which informs about the effects of bone microstructure on the piezoelectricity.

2:20

2pBA5. Relationships among ultrasonic and mechanical properties of cancellous bone. Keith A. Wear (Ctr. for Devices and Radiological Health, Food and Drug Administration, Bldg. 62, Rm. 2104, 10903 New Hampshire Ave., Silver Spring, MD 20993, keith.wear@fda.hhs.gov), Saghi Sadoughi (Univ. of California, Berkeley, CA), Srinidhi Nagaraja, Maureen Dreher (Ctr. for Devices and Radiological Health, Food and Drug Administration, Silver Spring, MD), and Tony M. Keaveny (Univ. of California, Berkeley, CA)

Most clinical bone sonometers measure broadband ultrasound attenuation (BUA) and speed of sound (SOS) in calcaneus. In addition, backscatter coefficient (BC) has been shown to have clinical utility. The objective of this work was to assess the extent to which ultrasonic measurements convey mechanical properties of cancellous bone. Twenty-five defatted human calcaneus samples were investigated *in vitro*. Normalized BUA (nBUA), SOS, and BC were measured using 500 kHz focused ultrasound transducers. Finite Element Analysis, based on micro-computed tomography images (Scanco microCT 100), was used to estimate stiffness and apparent modulus of the samples. Correlation coefficients from linear regressions were as follows: nBUA vs. stiffness—0.80 ($p < 0.0001$), nBUA vs. apparent modulus—0.81 ($p < 0.0001$), SOS vs. stiffness—0.80 ($p < 0.0001$), SOS vs. apparent modulus—0.84 ($p < 0.0001$), BC vs. stiffness—0.75 ($p < 0.0001$), and BC vs. apparent modulus—0.69 ($p < 0.001$). In conclusion, ultrasonic measurements are very sensitive to mechanical properties of cancellous bone. The mention of commercial products, their sources, or their use in connection with material reported herein is not to be construed as either an actual or implied endorsement of such products by the Department of Health and Human Services.

2:40–3:05 Break

3:05

2pBA6. Sample thickness dependence of Bayesian and modified least squares Prony's analysis methods on systematically shortened bovine cancellous bone. Amber Groopman (Phys., Washington Univ. in St. Louis, 1 Brookings Dr., Compton Hall, Campus Box 1105, St. Louis, MO 63130, nelsonam@wustl.edu), Keith Wear (U.S. Food and Drug Administration, Silver Spring, MD), Yoshiki Nagatani (Electronics, Kobe City College of Technol., Kobe, Japan), Katsunori Mizuno (Underwater Technol. Collaborative Res. Ctr., Univ. of Tokyo, Tokyo, Japan), Mami Matsukawa (Lab. of Ultrasonic Electronics, Doshisha Univ., Kyoto, Japan), Hirofumi Taki (Communications and Comput. Eng., Kyoto Univ., Kyoto, Japan), Jonathan Katz (Phys., Washington Univ. in St. Louis, St. Louis, MO), Mark Holland (Radiology & Imaging Sci., Indiana Univ. School of Medicine, Indianapolis, IN), and James Miller (Phys., Washington Univ. in St. Louis, St. Louis, MO)

The goal of this study was to compare the results from two proposed methods for separating fast and slow waves from mixed-mode signals, one using Bayesian probability theory and one using modified least-squares Prony's (MLSP) [Wear, J. Acoust. Soc. Am. **133**, 2490–2501 (2013)], on measurements of cancellous bone. Ultrasonic through-transmission data were acquired on a bovine femoral head specimen for thicknesses ranging from 15 mm to 6 mm. The thickness was reduced in 1 mm increments and measurements were acquired at each sample thickness [Nagatani *et al.*, Ultrasonics **48**, 607–612 (2008)]. A Bayesian parameter estimation analysis was performed on the experimentally acquired signals to isolate the fast and slow waves and to obtain estimates of the fast and slow wave's ultrasonic properties. Results for the corresponding ultrasonic properties were estimated using the modified least squares Prony's method plus curve fitting (MLSP + CF) on the same bovine sample by Wear *et al.* (J. Acoust. Soc. Am. **136**, 2015–2024). The results show good agreement between the phase velocities estimated by Bayesian and MLSP + CF analysis methods for both the slow wave and the fast wave. Both analysis methods yielded fast and slow wave phase velocities that depended on sample thickness. This may imply that the propagation model, which is used in both methods, may be incomplete.

3:25

2pBA7. Multiscale assessment of cortical bone properties with quantitative ultrasound. Johannes Schneider (Charité Universitätsmedizin Berlin, Berlin, Germany), Simon Bernard, Jean-Gabriel Minonzio (Laboratoire d'Imagerie Biomédicale, Paris, France), Peter Varga (AO Foundation, Davos, Switzerland), Robert Wendlandt (Biomechatronics, Lübeck, Germany), Quentin Grimal, Pascal Laugier (Laboratoire d'Imagerie Biomédicale, Paris, France), and Kay Raum (Charité Universitätsmedizin Berlin, Augustenburger Platz 1, Berlin 13353, Germany, kay.raum@charite.de)

Clinical bone quality assessment still relies strongly on bone mineral density (BMD). It is now well accepted that BMD predicts only around 60% of the individual fracture risk because it depends on geometrical dimensions of the bone and mechanical properties of its cortical shell. The tibia mid-shaft is a clinically relevant site for ultrasound axial transmission (AT) measurements, which allow simultaneous assessment of cortical thickness (Ct.Th) and tissue elasticity. This *ex-vivo* study investigated 19 representative human tibiae (native samples, mid-shaft, age range: 69–94 yrs) using AT with the aim to develop novel ultrasound biomarkers of cortical bone loss. $Ct.Th_{AT}$ was in good agreement with corresponding data from micro-computed tomography ($R2 = 0.90$). Resonant ultrasound

spectroscopy was used to measure the transverse isotropic elasticity tensor of extracted bone cubes. Strong linear correlation between axial stiffness (c33) and mass density ($R2=0.86$) was found. Mechanical testing based strength was moderately correlated to c33 ($R2=0.72$) and mass density ($R2=0.65$). These findings indicate the potential of cortical elasticity to be an adequate biomarker of bone quality which combined with information about bone geometry (e.g., Ct.Th) could improve clinical fracture risk assessment.

3:45

2pBA8. Backscatter difference techniques for bone assessment using an ultrasonic imaging system. Brent K. Hoffmeister, Morgan R. Smathers, Catherine J. Miller, Joseph A. McPherson, Cameron R. Thurston (Phys., Rhodes College, 2000 North Parkway, Memphis, TN 38112, hoffmeister@rhodes.edu), and Sang-Rok Lee (Health and Sport Sci., Univ. of Memphis, Memphis, TN)

Background: Backscatter difference techniques are being developed to detect changes in bone caused by osteoporosis. Backscatter difference techniques compare the power in one portion of an ultrasonic backscatter signal to the power in a different portion of the same signal. Goal: Evaluate the feasibility of using an ultrasonic imaging system to perform backscatter difference measurements of bone. Procedure: Ultrasonic signals and images were acquired from 24 specimens of bone using an ultrasonic imaging system (Terason) with a 5 MHz linear array transducer. The signals were analyzed to determine the normalized mean backscatter difference (nMBD) between two gated portions of each signal. The images were analyzed to determine the normalized pixel value difference (nPVD) between regions of interest (ROIs) positioned at two different depths. Results: nMBD demonstrated strong linear correlations with bone mineral density that ranged from 0.83 to 0.87. nPVD performed less well, yielding correlations that ranged from 0.42 to 0.81, depending on ROI separation. Conclusions: It is feasible to apply the backscatter difference technique to ultrasonic imaging systems for the purpose of bone assessment. Better results are obtained by analyzing the signals rather than the images.

4:05

2pBA9. Ultrasound backscattering from cancellous bone *in vivo*. Dean Ta, Tao Tang, Chengcheng Liu, and Weiqi Wang (Dept. of Electron. Eng., Fudan Univ., 220 Handan Rd., Shanghai 200433, China, tda@fudan.edu.cn)

Osteoporosis is a musculoskeletal disease characterized by a loss of bone mass and a deterioration of trabecular bone microarchitecture. In recent years, significant progress has been made in the quantitative ultrasound technique, and it has become well established as a noninvasive tool for the assessment of bone status. This report will show the ultrasonic backscatter theory in cancellous bone, analyze the ultrasonic backscattering signals, and to investigate correlations among backscatter parameters and bone mineral density (BMD) *in vivo*. Ultrasonic backscattering measurements were performed on 1226 subjects (601 males and 625 females) at the right calcaneus *in vivo* using the novel ultrasonic backscatter bone system. Then, the subjects underwent DXA for the BMD at sites located on the lumbar spine (sBMD) and left hip (hBMD). Spectral centroid shift (SCS), mean trabecular bone spacing (Tb.Sp), backscatter coefficient (BC), and apparent integral backscatter coefficient (AIB) were calculated at the central frequencies of 3.5 and 5.0 MHz. Linear regression showed that the SCS at 3.5 MHz exhibited negative correlations with the sBMD ($R=-0.66$) and hBMD ($R=-0.64$). The SCS at 5.0 MHz was also found to be closely related to the sBMD ($R=-0.68$) and hBMD ($R=-0.66$). Tb.Sp at 3.5 MHz and 5.0 MHz exhibited negative correlations with the sBMD ($R=-0.74$) and hBMD ($R=-0.66$).The correlations between backscatter parameters and BMD were found to be statistically significant in both the male and female groups.

4:25

2pBA10. Quantitative ultrasound imaging reconstruction in peripheral skeletal assessment. Yi-Xian Qin and Jesse Muir (Biomedical Eng., Stony Brook Univ., BioEng. Bldg., Rm. 215, Stony Brook, NY 11794, yi-xian.qin@stonybrook.edu)

Disuse osteopenia affect mineral density, microstructure and integrity of bone, which lead to increased risk of osteoporosis and fracture during long term space mission. In order to provide a non-ionizing, repeatable method of skeletal imaging, a novel hand-held scanning confocal quantitative ultrasound (QUS) device has been developed. A mobile scanning ultrasound images were collected using ten sheep tibia and eight human volunteers at the wrist, forearm, elbow, and humerus with the arm submerged in water. A custom MATLAB software used a least-error algorithm to adjust adjacent lines and remove unwanted image waver from hand movement. Ultrasound attenuation (ATT) values of 42.4 ± 0.6 dB and 41.5 ± 1.0 dB were found in water and gel coupling, respectively. Scans of the human subject revealed detailed anatomy of the humerus, elbow, forearm, and hand. Repeat measures of the distal radius found an attenuation of 37.2 ± 3.3 dB, with negligible changes cause by wrist rotation of $\pm10°$ (36.4 to 37.3 dB) indicating small changes in scan angle would not significantly affect bone quality measurements. The hand-held QUS device allows for rapid skeletal imaging without ionizing radiation. Information gathered from ultrasonic attenuation and velocity can provide detailed information on localized bone fragility as well as identify fractures.

Contributed Paper

4:45

2pBA11. Transcranial time-of-flight estimation using backscattered infrared data. Qi Wang, Mark Howell (Cleveland Clinic, 9500 Euclid Ave./ND 20, Cleveland, OH 44195, qiqiwang83@gmail.com), Namratha Reganti (Biomedical Eng., Case Western Reserve Univ., Cleveland, OH), and Gregory T. Clement (Cleveland Clinic, Cleveland, OH)

Recently, we reported on our observation that diffuse infrared (IR) light transmitted through human skull bone exhibits an intensity that correlates positively with acoustic (1 MHz) time-of-flight data acquired at equivalent locations [POMA **22**, 020002 (2015)]. Presently, we investigate the potential to exploit this correlation as a means of correcting for skull-induced aberration. For measurement, a dual ultrasonic/infrared array capable of illuminating a localized region of the skull bone and recording the backscattered light was constructed. The array design utilized 940 nm IR emitters (TSAL4400, 3-mm-diameter) and a single detector (TEFD4300-3 mm) configured to transmit and record under the face of an ultrasonic PZT ring transducer. Initial tests consisted of a three-emitter configuration, with fiberglass-reinforced foil used to isolate the direct infrared signal between the emitters and the receiver. Effects of skin were considered by attaching a skin-mimicking layer to the skull and comparing results with and without the layer. Data acquired on a set of eight skull samples will be presented. Preliminary results show good correlation between ultrasound and IR intensity. [Work supported by NIH award R01EB014296.]

Posters will be on display and contributors will be at their posters during the session break from 2:40 p.m. to 3:05 p.m.

Invited Paper

2pBA12. Numerical investigation of fast and slow longitudinal waves backscattered from various depths inside cancellous bone.
Atsushi Hosokawa (Dept. of Elec. and Comput. Eng., National Inst. of Technol., Akashi College, 679-3 Nishioka, Uozumi, Akashi 674-8501, Japan, hosokawa@akashi.ac.jp)

Ultrasound backscattering in cancellous bone was numerically investigated using three-dimensional (3D) finite-difference time-domain (FDTD) simulations with microcomputed tomographic (μCT) models of the bone. The cancellous bone models with various thicknesses, in which artificial absorbing layers were set at the surfaces opposite to the ultrasound-transmitted surfaces, were prepared. The backscattered waveform inside cancellous bone was isolated by deriving the difference between two simulated waveforms obtained using the bone models with different thicknesses, and the backscatter properties from various depths of the bone were investigated. When an ultrasound pulse wave was transmitted in the direction parallel to the main orientation of the trabecular network, the backscattered waves from the deep bone depths could clearly separate into two waves, which could correspond to the backscattered waves of the fast and slow longitudinal waves propagating mainly in the trabecular elements and the pore spaces. For each of the fast and slow waves, the backscatter coefficient, which was the ratio of the backscattered fast/slow wave amplitude to the incident fast/slow wave amplitude, was obtained. The results showed that both backscatter coefficients were weakly correlated with the bone depth.

Contributed Papers

2pBA13. Broadband ultrasound scanning and frequency sweep measurement for trabecular bone with novel wideband single crystal transducer.
Jian Jiao, Xiaofei Li, Liangjun Lin (Biomedical Eng., Stony Brook Univ., 100 Nicolls Rd., 212 BioEng. Bldg., Stony Brook, NY 11794, jian.jiao@stonybrook.edu), Raffi Sahul, Ed Nesvijski (TRS Technologies Inc., State College, PA), and Yi-Xian Qin (Biomedical Eng., Stony Brook Univ., Stony Brook, NY)

Quantitative ultrasound has been developed as a radiation-free technique to evaluate both bone density and strength. The ultrasound parameters, such as velocity (UV), attenuation (ATT), and broadband attenuation (nBUA), have been widely used as indicators for bone health status. In our study, a novel spiral-wrapped wideband ultrasound transducer is developed by TRS. Bovine trabecular bone samples (15 mm thickness) were prepared for frequency scanning. In each group, TRS transducer was used to emit the frequency from 0.5 to 7 MHz to propagate through samples; and pulses were received by the hydrophone. Frequency sweep response mostly correlated to bone density at frequency under 3 MHz, and barely showed any correlation with bone density at a frequency above 3 MHz, as higher frequency ultrasound energy attenuated significantly in trabecular bone. With controlled bone density reduction by 34.71%, UV decreased from 2575.03$\pm$34.38 m/s to 1717.15$\pm$48.19 m/s, ATT decreased from 3.66$\pm$0.71dB to 2.21$\pm$0.76 dB, and nBUA decreased from 2.09$\pm$0.96 dB/MHz/cm to 1.21$\pm$0.47 dB/MHz/cm. Generally, bone density decrease resulted in UV, ATT, and nBUA decreases, but were sensitive to the change of the wideband frequency from 0.5 MHz to 0.8 MHz. This new wideband transducer offers more information than the conventional ultrasound transducer, and can thus be used as a modality to evaluate bone health status.

2pBA14. Ultrasonic wave velocities in radial direction of bovine cortical bone.
Yuma Nishimura (Faculty of Sci. and Eng., Doshisha Univ., 1-3 Tatara-Miyakodani, Kyotanabe, Kyoto 610-0321, Japan, duo0347@mail4.doshisha.ac.jp), Satoshi Kawasaki (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Kyoto, Japan), and Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan)

The ultrasonic axial transmission (AT) techniques have been proposed to evaluate the cortical bone of arms and legs. For the mode analysis of propagating waves, the information of anisotropy and heterogeneity of the cortical bone is important. Yamato has pointed that longitudinal wave velocity in the axial direction of bovine cortical bone changes due to the measurement positions and microstructures. In this study, we have investigated longitudinal wave velocities in radial direction of cortical bone. The cuboid samples were obtained from the cortical bone of bovine femur. During ultrasonic pulse immersion experiments, a sinusoidal ultrasonic wave in the range from 3 to 10 MHz was transmitted from a PVDF transducer, which passed through the sample in the radial direction. By reducing the sample thickness gradually, wave velocities were measured. The average velocities in radial direction of samples were around 3450 m/s and depended on the positions of the samples in the femur. They variation of the velocities due to the thickness was about 7 %. These data give us the information of heterogeneity of cortical bone in radial direction which should be considered in the AT techniques. [1] Y. Yamato, *et al*., Jpn. J. Appl. Phys., **47**, 4096–4100 (2008).

2pBA15. FDTD simulations of ultrasonic wave propagation in the cortical bone with heterogeneity.
Toshiho Hata (Faculty of Life and Medical Sci., Doshisha Univ., 1-3 Tatara, Miyakodani, Kyotanabe, Kyoto 610-0394, Japan, dmo1008@mail4.doshisha.ac.jp), Yoshiki Nagatani (Kobe City College of Technol., Kobe, Japan), and Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan)

The ultrasonic axial transmission (AT) technique is known to assess the cortical bone of femur and tibia. After the first application of the finite difference time domain (FDTD) method [1], the wave propagation in bone has been discussed by both simulation and experimental studies. The wave propagation depends on the wave velocity, bone geometry (curvature, cortical thickness), anisotropy, heterogeneity, etc. In this study, we have investigated the ultrasonic wave propagation in the cortical bone with heterogeneity (stiffness distribution) by FDTD method. Using the stiffness distribution of bovine cortical bone, experimentally measured by Yamato and Nakatsuji [2], [3], wave propagation of one cycle of sinusoidal wave at 0.5~1 MHz was investigated, assuming cuboid samples. The wave velocities inside the bone were higher than those of the bone surfaces. The complex propagating waves were slightly concentrated at the bone surface. The results indicate that the stiffness distribution is also an important factor to understand the wave propagation. [1] E. Bossy *et al*., J. Acoust. Soc. Am. **115** , 2314 (2004). [2] Y. Yamato *et al*., Jpn. J. Appl. Phys. **47**, 4096 (2008). [3] T. Nakatsuji *et al*., Jpn. J. Appl. Phys. **50**, 07HF18 (2011).

Session 2pED

Education in Acoustics: Topics in Acoustics Education

Andrew A. Piacsek, Chair
Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926

Contributed Papers

1:30

2pED1. Acoustic engineering by design—Example of a service learning course for undergraduate engineering students. Eoin A. King, Philip P. Faraci, and Robert D. Celmer (Acoust. Prog. and Lab, Mech. Eng., Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, eoking@hartford.edu)

The University of Hartford has two ABET-accredited undergraduate engineering programs in the area of acoustics—a Bachelor of Science in Mechanical Engineering with acoustics concentration and a Bachelor of Science in Engineering with a major in acoustical engineering and music. Students participating in both of these programs take an acoustic engineering design course with a focus on service learning during their sophomore year. Service learning is an alternative to the traditional teaching model and offers students the opportunity to apply their classroom studies to local issues in the community; thus, students achieve academic objectives by meeting real community needs. Each year members of the community approach the University to request assistance with rooms exhibiting poor acoustic characteristics due to improper reverberation times and/or excessive background noise levels. Students are required to design and communicate solutions to these issues in a manner similar to a professional consultation. Recent examples include a science center exhibit hall, a restaurant, an architecture studio classroom, and a retirement community auditorium. This presentation will provide examples of such projects completed as part of this class. Challenges and benefits associated with this service learning course for acoustic engineering undergraduate students will be discussed.

1:45

2pED2. Development of an acoustics outreach program for the deaf. Cameron T. Vongsawad, Mark L. Berardi, Kent L. Gee, Tracianne B. Neilsen, and M J. Lawler (Phys. & Astronomy, Brigham Young Univ., 1041 E. Briar Ave., Provo, UT 84604, cvongsawad@byu.net)

The Hear and Seemethodology (Groppe, 2011) has often been used as a means of enhancing pedagogy by focusing on the two strongest learning senses, but this naturally does not apply to deaf or hard of hearing students. Because deaf students' prior nonaural experiences with sound will vary significantly from those of students with typical hearing, different methods must be used to build understanding. However, the sensory-focused pedagogical principle can be applied in a different way for the Deaf by utilizing the senses of touch and sight, called here the "See and Feel" method. This presentation will provide several examples of how acoustics demonstrations have been adapted to create an outreach program for a group of junior high students from a school for the Deaf and discuss challenges encountered.

2:00

2pED3. Using acoustics and vibrations to teach experimental concepts and technical communications. Teresa J. Ryan (Dept. of Eng., East Carolina Univ., 248 Slay Hall, Greenville, NC 27858-4353, ryante@ecu.edu), Diego Turo, Joseph F. Vignola, and John A. Judge (Mech. Eng., The Catholic Univ. of America, Washington, DC)

This work describes an experience-based conceptual course in vibrations and acoustics. This one-hour course is designed with the goal of framing general experimental issues, interpreting experimental results, and developing technical communication skills in engineering by way of acoustics and vibrations experiments. Like other laboratory experiences, the goal is to support the skill of making the connection between experimental results and the physical system. Unlike other courses offered at East Carolina University and The Catholic University of America, the primary learning objectives are those connections and technical communication skills rather than the acoustics and vibrations content. Experimental issues explored include error, repeatability, linearity, uncertainty, and general experimental design. This particular set of acoustics and vibrations experiments is well suited to this purpose because they do not require extensive laboratory infrastructure, they are inherently safe, and they use commonplace tools. These tools include a speaker, microphone, and accelerometer, which the students already have as part of their laptop and/or smartphone. These tools, along with a copy of MATLAB, provide the opportunity to perform a wide array of experiments, which can illustrate sophisticated concepts using these basic, familiar items.

2:15

2pED4. A simply constructed condenser microphone as a teaching aid. Randall A. Ali (Dept. of Elec. and Comput. Eng., Univ. of the West Indies, Eastern Main Rd., St. Augustine na, Trinidad and Tobago, randall.ali@sta.uwi.edu) and Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

To gain a deeper appreciation and a better understanding of the operation of condenser microphones, it is instructive to have a tangible device, which can undergo a fair amount of experimentation. Low cost condenser microphones used in laptops, mobile phones, and other similar devices can be purchased, but not necessarily deconstructed to serve as an effective learning tool. On the other hand, condenser microphones used in recording studios can be taken apart, but can become quite expensive and making any elaborate modifications to the design may prove challenging. As an alternative, we present a functional and cost-effective condenser microphone that can be simply constructed from readily available materials. The device enables students to make changes to the diaphragm material, back plate, electrostatic gap spacing, and the electronic circuit for connection to a lower impedance device. A SPICE model of the condenser microphone is also presented, which was used to guide the design and can provide further insight for the student.

2:30

2pED5. Impedance tube experiments as a tool to teach acoustics of porous media. Diego Turo, Aldo A. Glean, Chelsea E. Good, Joseph F. Vignola (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., N.E., Washington DC, 20064, diegoturo@gmail.com), Teresa Ryan (Eng., East Carolina Univ., Greenville, NC), and John A. Judge (Mech. Eng., The Catholic Univ. of America, Washington DC, DC)

As a practical matter, sound propagation in many everyday situations includes interactions with porous materials. Such materials include textiles, foams, gravel, grass-covered ground, asphalt, and trees. Modeling sound propagation in complex environments, such as passenger cabins or urban spaces, with porous materials like these is essential to understanding realistic acoustics problems. The theory behind the sound absorption in porous material is not typically included in introductory acoustics courses. This is, at least in part, attributable to the mathematical complexity of the topic. Despite this barrier, acoustic properties of porous material can be easily introduced to typical undergraduate students using simple experiments to demonstrate the acoustic effects of different materials. This presentation describes acoustic property measurements of rigid-frame porous material with straight cylindrical pores as well as foam and soil included in an "Introduction to Acoustics" class at the Catholic University of America. The apparatus is composed of a LabVIEW measurement interface, a pair of microphones, and a simple aluminum impedance tube. The modular design allows students to adapt the software or hardware for other experiments. These hands-on experiences are intended to impart a conceptual understanding of the theory of sound absorption and the acoustic performance of sound absorbing (porous) media.

2:45

2pED6. Energy flux streamlines versus acoustic rays for modeling an acoustic lens: Testing and refinement. Cleon E. Dean (Phys., Georgia Southern Univ., PO Box 8031, Math/Phys. Bldg., Statesboro, GA 30461-8031, cdean@georgiasouthern.edu) and James P. Braselton (Mathematical Sci., Georgia Southern Univ., Statesboro, GA)

As an extension of recently published experimental work [Cleon E. Dean and Kendez Parker, "A ray model of sound focusing with a balloon lens: An experiment for high school students," J. Acoust. Soc. Am. **131**, 2459–2462 (2012)], preliminary results comparing energy flux streamlines [David M. F. Chapman, "Using streamlines to visualize acoustic energy flow across boundaries," J. Acoust. Soc. Am. **124**, 48–56 (2008)] versus acoustic rays for visualizing the energy flow inside and in the focal region of an acoustic lens in the form of a carbon dioxide filled balloon in air were presented. The sound field was expanded in the usual Legendre polynomials and spherical Bessel functions, and the energy flux vectors at points throughout the regions of interest were calculated [J. Adin Mann III, et al., "Instantaneous and time-averaged energy transfer in acoustic fields," J. Acoust. Soc. Am. **82**, 17–30 (1987)]. Deficiencies in the streamline plotting routines used in this earlier version of Mathematica and subtle differences between acoustic rays and acoustic energy flux streamlines lent itself to an inaccurate perception of the results. This talk uses Mathematica 10 routines to revisit these results and concentrates on testing and verification of the conclusions from the previous work.

3:00

2pED7. Development of boundary element method to analyze acoustic models with comparison to finite element model. Mahdi Farahikia and Ronald N. Miles (Mech. Eng., SUNY Binghamton, 13 Andrea Dr. Apt. A, Vestal, NY 13850, mfarahi1@binghamton.edu)

Numerical and computational models provide valuable information about engineering designs. Of their desirable benefits is reduction of expenses and feasibility of observing the change in behavior of a model with varying design parameters. In this paper, a boundary element method that was developed in Matlab is described. The approach in developing this method is based on the Green's function for an unbounded acoustical domain. This model allows reading of computer aided designs to further analyze them in acoustic domain. The results obtained from this method are compared with equivalent finite element models carried out in ANSYS and exact analytical solutions to check for accuracy. Steps in analyzing acoustic models in ANSYS and Matlab are also explained in this paper.

TUESDAY AFTERNOON, 19 MAY 2015 — KINGS 5, 1:30 P.M. TO 2:55 P.M.

Session 2pNSa

Noise and Psychological and Physiological Acoustics: Application of Psychoacoustics in Noise

Klaus Genuit, Cochair
HEAD Acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany

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

Invited Papers

1:30

2pNSa1. Environmental noise assessment by means of psychoacoustics and its relation to *in-situ* annoyance judgments of urban noise. André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, andre.fiebig@head-acoustics.de)

It is obvious that environmental noise assessment only based on the consideration of L_{Aeq} values cannot determine the character of environmental noise and its specific reactions and responses evoked. Noise is not only perceived in terms of lower or higher (averaged) sound pressure levels; noise leads to different auditory sensations and perceptions, which are influenced by context. The introduction of psychoacoustics allows for describing the character of noise more in detail and for predicting human responses to noise more reliably. In order to investigate the link between psychoacoustic parameters and annoyance judgments, soundwalk data including *in-situ* noise judgments from several measurement campaigns were analyzed. It turned out that by taking into account psychoacoustic parameters more variance in noise annoyance data is explained. Based on the analysis and results of the soundwalk data, the questions are addressed about how long must be measured to encompass all relevant acoustical situations and how the measurement of short-term reactions are related to long-term noise effects. Based on the presented case studies, it is intended to gain insight into the benefits and limitations of psychoacoustics with respect to environmental noise assessment.

1:50

2pNSa2. Status of psychoacoustics in noise analysis. Klaus Genuit (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath, NRW 52134, Germany, Klaus.Genuit@head-acoustics.de)

Psychoacoustics provides knowledge about the relationship between acoustical stimuli and provoked hearing sensations for specific "one-dimensional" sensations like loudness, sharpness, roughness, or fluctuation strength. Psychoacoustic measures are often applied in the context of sound quality investigations. Sound quality has to be understood as a multi-dimensional phenomenon related to the perception of sensory pleasantness (sound character) and suitability of sound in context. It is widely accepted that psychoacoustic measures offer better agreement with auditory sensation than conventional A-weighted sound pressure levels and spectra do. In particular, the psychoacoustic parameter loudness gains extensively in significance, because it shows a higher correspondence with the sensation of volume (loudness) than any sound level indicator. Thus, loudness represents a dominant feature for sound quality evaluation and is frequently applied in numerous applications. In the field of *environmental noise* the main focus lies on the exact measurement and description of the acoustical situation in a pure physical sense, whereas the community noise perspective tries to bridge acoustical exposure and the human assessment of noise in sense of annoyance level. In contrast to it, the disciplines of psychoacoustics, sound quality, and soundscape put more emphasis on the perceiving human being. This paper gives an overview about the status of psychoacoustics with respect to application in noise.

Contributed Paper

2:10

2pNSa3. Tuning the cognitive environment: Sound masking with "natural" sounds in open-plan offices. Alana G. DeLoach (School of Architecture (Architectural Acoustics), Rensselaer Polytechnic Inst., PO Box 413, Nantucket, MA 02554, deloaa3@rpi.edu), Jeff P. Carter, and Jonas Braasch (School of Architecture (Architectural Acoustics), Rensselaer Polytechnic Inst., Troy, New York)

With the gain in popularity of open-plan office design and the engineering efforts to achieve acoustical comfort for building occupants, a majority of workers still report dissatisfaction in their workplace environment. Office acoustics influence organizational effectiveness, efficiency, and satisfaction through meeting appropriate requirements for speech privacy and ambient sound levels. Implementing a sound masking system is one tried-and-true method of achieving privacy goals. Although each sound masking system is tuned for its specific environment, the signal—random steady state electronic noise, has remained the same for decades. This session explores how "natural" sounds may be used as an alternative to this standard masking signal employed so ubiquitously in sound masking systems in the contemporary office environment. As an unobtrusive background sound, possessing the appropriate spectral characteristics, this proposed use of "natural" sounds for masking challenges the convention that masking sounds should be as meaningless as possible. Based on psychophysical data and a sound-field analysis through an auditory model, we hypothesize that "natural" sounds as masking sounds have the ability (with equal success as conventional masking sounds) to meet standards and criteria for speech privacy while enhancing cognitive functioning, optimizing the ability to concentrate, and increasing overall worker satisfaction.

2:30–2:55 Panel Discussion

Session 2pNSb

Noise and Psychological and Physiological Acoustics: Mobile Technology Solutions for Hearing Loss Prevention

William J. Murphy, Cochair
Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Institute for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998

Michael G. Heinz, Cochair
Speech, Language, and Hearing Sciences & Biomedical Engineering, Purdue University, 500 Oval Drive, West Lafayette, IN 47907

Chair's Introduction—3:30

Invited Papers

3:35

2pNSb1. Accurate mobile noise measurements: Challenges for the app developer. Benjamin Faber (Faber Acoust., LLC, 277 S 2035 W, Lehi, UT 84043, ben@faberacoustical.com)

The ubiquity of mobile computing devices, their connection to the cloud, and their media-centric nature makes them attractive for wide scale acquisition of acoustical information, including human noise exposure in various environments and at various times. One concern in harnessing the potential of such a network of devices is ensuring that each device is able to acquire the desired acoustical information with sufficient accuracy. It is possible to make accurate acoustical measurements, as long as care is taken to ensure that the hardware, software, and users all work together to ensure success. System software, signal acquisition hardware, microphones, the mobile devices and the people who use them each present certain issues and challenges to the process of making accurate and reliable acoustical measurements. Various of these issues and challenges will be presented and discussed from an app developer's perspective.

3:55

2pNSb2. Use of smartphone sound measurement apps for occupational noise assessments. Chucri A. Kardous (National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Cincinnati, OH 45226, ckardous@cdc.gov) and Metod Celestina (EA LAB, Ljubljana, Slovenia)

NIOSH conducted two studies to examine the accuracy of smartphone sound measurement applications (*apps*). The first study examined 192 sound measurement *apps* on the Apple (iOS) and Google (Android) platforms. Only 10 iOS *apps* met our selection criteria for functionality, measurement metrics, and calibration capability. The studies compared the performance of the *apps* with a reference microphone and with a professional type 1 sound level meter and a type 2 noise dosimeter. The results showed 4 iOS *apps* with means of differences within $\pm$ 2 dB(A) of the reference microphone. The Android-based *apps* lacked the features and functionalities found in iOS *apps* and showed a wide variance between the same *app* measurements on different devices. A follow-up study of the 4 iOS *apps* using calibrated external microphones (MicW i436 and Dayton Audio iMM-6), showed an even closer agreement with professional meters. Overall, the studies suggest that certain *apps* may be used for some occupational noise assessments but only if properly calibrated and used within the hardware limits of the mobile devices. NIOSH and EA LAB are collaborating to develop an occupational sound measurement *app* for iOS devices in an effort to improve awareness of the noise hazards in the workplace.

4:15

2pNSb3. Acoustic requirements for audiometric testing and hearing protector fit-testing with mobile platforms. William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Mailstop C-27, Cincinnati, OH 45226-1998, wjm4@cdc.gov)

Mobile technology platforms present new challenges for implementing hearing loss prevention programs. Laptop computers, smartphones, and tablets have capabilities to generate 16-bit and sometimes 24-bit audio stimuli for hearing tests. For testing hearing at threshold, the noise floor of the sound card of the computer can pose a limitation. During the development of sound compression standards (e.g., MPEG, AAC, MP3), 24-bit resolution and 96 kHz sampling is necessary to encompass the requirements for dynamic level the frequency range of human hearing. However, hearing screening and fit-testing can be accomplished with 44 kHz sampling rates. However, the bit resolution deserves careful treatment to ensure that output is accurate. This paper will describe features for mobile applications for audiometric testing and hearing protector fit-testing. Several external DAC platforms on Windows 7 based laptops along with a small sample of iOS and Android devices have been tested and demonstrate differing levels of performance. Linearity of the systems and output levels for a standard audiometric headphone will be presented.

4:35

2pNSb4. Comparison of headphones for audiometric screening and hearing protector fit-testing. David C. Byrne (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, 1090 Tusculum Ave., Cincinnati, OH 45226-1998, zne2@cdc.gov), Laura Schmitt (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), and William J. Murphy (Hearing Loss Prevention Team, Centers for Disease Control and Prevention, National Inst. for Occupational Safety and Health, Cincinnati, OH)

Accurate hearing screening in hearing loss prevention programs is entirely dependent upon sufficient isolation of the subject from any surrounding noise as well as adequate acoustical performance of the headphones that are used. Assessment of hearing protector attenuation by newly developed individual fit-test systems uses many elements borrowed from standard audiometric testing. To assess the suitability for this type of testing, the National Institute for Occupational Safety and Health evaluated the attenuation characteristics, maximum output levels, linearity, and total harmonic distortion of three different circumaural headphones: the FitCheck headphone with a TDH-39 driver, the Sennheiser HDA-200, and Sennheiser HDA-300 headphones. Attenuation measurements were obtained with a laboratory-based real-ear attenuation at threshold system, and the electroacoustic measurements were conducted on a standard IEC318 coupler. The long term goal of this effort is to determine whether attenuation differences or the performance characteristics of the different styles of headphones affect the resulting earplug attenuation values when measured with an earplug fit-test system.

Contributed Paper

4:55

2pNSb5. Characterization of the smart-phone radiated noise using near-field acoustic holography. Donghyun Kim, Taeyoung Park, and Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., Eng. Bldg. A391, 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, kimdh88@yonsei.ac.kr)

Smart phones have dramatically evolved on the functional and aesthetic fronts in recent years. However, as smart phones get slimmer, problems arise where the noise radiated from the smart phone, albeit small in SPL, interferes with the conversation and degrades the call quality. The noise radiation is primarily due to the vibration of the internal electronics components and its transmission through the circuit board and the casing. In this paper the nearfield acoustic holography (NAH) is used to characterize the 3-D sound field of the smart-phone radiated noise, which is tailored specifically to deal with the small size and the low SPL of the sound radiator. Quantities such as acoustic pressure, particle velocity, and intensity at the smart phone surface are obtained and utilized to identify parts that are responsible for noise generation. In this respect, the NAH proves to be an effective tool for the characterization and control of the smart phone acoustics.

TUESDAY AFTERNOON, 19 MAY 2015 KINGS 1, 1:00 P.M. TO 4:00 P.M.

Session 2pPA

Physical Acoustics: General Topics in Physical Acoustics II

Kevin M. Lee, Chair

Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78758

Contributed Papers

1:00

2pPA1. Ultrasound propagation in concentrated random dispersions of spherical particles: Thermal- and shear-mediated contributions to multiple scattering. Valerie Pinfield and D Michael Forrester (Chemical Eng. Dept., Loughborough Univ., Loughborough LE11 3TU, United Kingdom, v.pinfield@lboro.ac.uk)

Ultrasonic techniques offer advantages for process monitoring for dispersions of colloidal or nano particles; such materials occur in a wide variety of process industries. However, the application of ultrasonic techniques has been limited by the inaccuracy of ultrasonic propagation models used to interpret the measurements (typically attenuation spectra). Multiple scattering models, such as the Lloyd and Berry model [Proc. Phys. Soc. London **91**, 678 (1967)], have been used with great success in relatively dilute colloidal dispersions, but fail for higher concentrations, smaller particles, and low frequencies, primarily due to the neglect of thermal- and shear-mediated effects. We present a modified multiple scattering model that includes these thermal- and shear-wave contributions and explore their significance. The model develops work by Luppé, Conoir and Norris [J. Acoust. Soc. Am. 2012 (131) 1113) for compressional, thermal and shear wave propagation. We identify the dominant scattering contributions for emulsions [Pinfield, J. Acoust. Soc. Am. **136**, 3008 (2014)] and suspensions and develop analytical forms for them. Numerical calculations demonstrate the contribution of the additional multiple scattering effects to the compressional wave speed and attenuation through the emulsion or suspension. The calculations are compared with previously published experimental data.

1:15

2pPA2. Direct visualization of shear waves in viscoelastic fluid using microspheres. Cecille Labuda (National Ctr. for Physical Acoust., Univ. of MS, 1 Coliseum Dr., University, MS 38677, cpembert@olemiss.edu), Connor M. Tierney, E. G. Sunethra K. Dayavansha (Phys. and Astronomy, Univ. of MS, University, MS), and Joseph R. Gladden (National Ctr. for Physical Acoust., Univ. of MS, University, MS)

Wormlike micellar fluids are viscoelastic and can thus support shear waves. Shear waves in 500 mM CTAB-NaSal micellar were visualized by seeding the fluid with microspheres. This method was compared to visualization by observation of birefringence patterns induced by fluid strain in response to shear stresses. Shear speeds measured using both techniques were consistent. Particle displacement was observed to be a sinusoidal function of time and displacement amplitude decreased quadratically with distance from the source. This supports the possibility of using particle amplitude measurements as a measure of attenuation even at low fluid concentration where birefringence visualization techniques fail.

1:30

2pPA3. Determination of the phase speed in a bubbly liquid at the single bubble resonance in an impedance tube using a transfer function technique. Stanley A. Cheyne, Hugh O. Thurman, Walter C. McDermott, and Charles G. Kelley (Dept. of Phys. & Astronomy, Hampden-Sydney College, Hampden-Sydney, VA 23943, scheyne@hsc.edu)

The Transfer function method was used to determine the phase speed in a bubbly liquid at the single bubble resonant frequency. The Transfer function technique is widely used to determine the impedance of an absorbing sample. Once the impedance of the sample is found, the phase speed can be determined. White noise was generated and allowed to propagate upward in a vertical, steel, impedance tube. The top was terminated by a bubble cloud generated by pressurized air forced through hypodermic needles. The sound spectrum was measured at two different points in the pure water by carefully moving a single hydrophone. The transfer function was then determined and used to calculate the phase speed in the bubbly liquid.

1:45

2pPA4. Acoustic measurements and modeling of air-filled underwater resonator cavities. Laura M. Tseng, Kevin M. Lee (Appl. Res. Laboratories: The Univ. of Texas at Austin, 10,000 Burnet Rd., Austin, TX 78758, ltseng@utexas.edu), and Preston S. Wilson (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

This project investigated the near-resonance acoustical properties of submerged air-filled resonator cavities intended for use in underwater noise abatement systems, exploring a potential alternative to encapsulated bubbles. The underwater resonator cavities are air-filled containers resembling Helmholtz resonators, inverted and submerged in water, creating a direct interface between the volume of air trapped inside the cavity and the surrounding water. Experiments were performed with a prototype resonator cavity in a closed water-filled laboratory tank operated in the long wavelength limit (under 500 Hz) and the resonance frequencies and Q-factors were measured for various air-fill volumes. A finite-element simulation was used to examine the behavior of the resonator cavities within the experimental tank apparatus, yielding good agreement with measurements. Finally, free-field models were also developed (a finite-element numerical model and a Helmholtz-resonator-based analytical model), both of which yielded good agreement with the tank measurements. The measurements and models presented here indicate that inexpensive and convenient sub-wavelength laboratory tank measurements, along with simple analytical models can be used to accurately design and verify the free-field behavior of low frequency underwater noise abatement resonators.

2:00

2pPA5. Acoustic emissions of ice fracture. Melisa Yashinski, Rintaro Hayashi, Tyler Heei-Wai, and John S. Allen (Mech. Eng., Univ. of Hawaii, 2540 Dole St., Holmes Hall 302, Honolulu, HI 96822, alleniii@hawaii.edu)

Acoustic signals of ice movement and breaking has been recorded in previous field studies, but the specific role of salinity on ice fracture noise remains an open research topic. In lab studies, salinity has been shown to alter the mechanical properties but the associated emissions are not well understood. In this study, high-speed optical visualization is done in conjunction of acoustic measurements of ice fracture for three salinity values in air and underwater. Fracture strength is determined through three point bending and the role of columnar structure representative of Arctic ice is explored. The elastic–plastic behavior is characterized through time-frequency analysis.

2:15

2pPA6. Direct computation of acoustic scattering and difference frequency generation. Chrisna Nguon (Univ. of Massachusetts Lowell, 63 Hemlock St., Dracut, MA 01826, chrisna_Nguon@student.uml.edu), Ololade Mudasiru, Sameera Mogulla, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, Lowell, MA)

A simulation of the scattered field at the difference-frequency resulting from the nonlinear interaction of two beams at differing frequencies incident on a fluid scatterer is performed. The total wavefield response is modeled using a finite-difference time-domain approximation and a perfectly matched layer to estimate the spatial scattering features of the medium. A pulsed ultrasonic transducer beam is modeled and its interaction with a fluid scatterer with spatially varying compressibility contrast is visualized. The second-order pressure field is determined as a response to the difference-frequency source field constructed from the first order-response of the scatterer to the two incident beams as well as a spatial distribution of the nonlinear parameter. The performance and accuracy of the numerical simulation is examined.

2:30–2:45 Break

2:45

2pPA7. Evaluation of the spatial impulse response of planar ultrasonic radiators. Ayse Kalkan-Savoy, Nicholas Misiunas, J. Cory Miniter, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, University of Massachusetts Lowell, One University Ave., CACT FA203, Lowell, MA 01854, ayse.k.savoy@gmail.com)

A method for evaluating the spatial impulse response for a planar source having arbitrary velocity and delay distribution is presented. The time-domain approach is based on the Rayleigh integral formulation for the velocity potential. To obtain the velocity potential one must evaluate and superpose the contribution from each elemental element of the radiator. This step entails the accumulation of the amplitude contribution from each elemental source arriving at the observation point at a prescribed time. The problem of sorting of these arrivals is examined in this work and procedures for mapping the algorithm for implementation on general-purpose graphics processing units are presented.

3:00

2pPA8. The weak sensitivity of acoustic resonance frequencies to temperature gradients. Keith A. Gillis, Michael R. Moldover (Sensor Sci. Div., National Inst. of Standards and Technol., 100 Bureau Dr., Mailstop 8360, Gaithersburg, MD 20899-8360, keith.gillis@nist.gov), James B. Mehl (Dept. of Phys. and Astronomy, Univ. of Delaware, Orkas, WA), and James W. Schmidt (Sensor Sci. Div., National Inst. of Standards and Technol., Gaithersburg, MD)

We determined the mass of argon gas contained within an un-thermostatted, commercially-manufactured 300 L pressure vessel (tank) with an uncertainty of 0.16 % at 0.6 MPa by combining measurements of the argon pressure, the frequencies of microwave and acoustic resonances within the tank, and an equation of state for argon. After correction for the

thermoacoustic boundary layer and for the tank's center-of-mass motion, the measured acoustic resonance frequencies f_{ac} determined the average speed of sound, and therefore the average temperature, of the argon in the tank. We show that, consistent with first-order perturbation theory, f_{ac} for the 3 lowest-frequency longitudinal gas modes gave the correct average temperature even when we imposed a 13 K temperature difference ΔT across the tank's diameter. However for the nearly degenerate doublet modes, we observed a linear dependence on ΔT for f_{ac}, which the theory does not predict. Using the thermal expansion deduced from the microwave measurements, we show that the linear dependence on ΔT was consistent with anisotropic changes in the tank's shape in response to the applied temperature gradient. We will discuss the predictions from perturbation theory of the changes in f_{ac} due to temperature gradients in cylindrical and spherical cavities. From these results, we argue that resonance frequencies can be used to "weigh" a compressed gas in much larger tanks in un-thermostatted environments and at high pressures.

3:15

2pPA9. Absorption and dispersion in Venus' lower and middle atmosphere. Mathbar S. Raut and Andi Petculescu (Univ. of Louisiana at Lafayette, P.O. Box 44210, Lafayette, LA 70504, msr0706@louisiana.edu)

Vertical profiles for the sound speed and attenuation coefficient are calculated in the troposphere and mesosphere of Venus. The predictions are based on ambient quantities (pressure, temperature, density, and composition) obtained by measurements and global circulation models. The three major species considered for composition are carbon dioxide, nitrogen, and sulfur dioxide. The thermophysical parameters—specific heats, viscosity, and thermal conductivity—for the principal constituents were obtained by interpolation from the NIST Chemistry Webbook, at the pressure/temperature values of each altitude. A real-gas equation of state was used to account for the dense environment of the Venusian troposphere.

3:30

2pPA10. Investigation of acoustic scattering using fast-multipole Padé approximant methods. Hui Zhou, Elaheh Noursadeghi, Aydin Sadeqi, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, One University Ave., CACT FA203, Lowell, MA 01854, hui_zhou@student.uml.edu)

The scattering of acoustic waves from three-dimensional compressible fluid scatterers is considered. Particular attention is paid to cases where the scatterers have moderate magnitude in compressibility contrast and nondimensional wave number. The scattered field is cast in terms of asymptotic series valid in the limit of small compressibility contrast. It has been shown that pointwise application Padé Approximants in the scattering volume may be used to expand the range of validity of the result. However pointwise divergence in the scattering volume can result as the magnitude of the compressibility contrasted is increased. In this work the utility of low-frequency Fast-multipole method in evaluating the Born series coefficients is examined.

3:45

2pPA11. Accurate fast field formulation for a uniformly moving source traveling above an impedance plane. Bao N. Tong and Kai Ming Li (School of Mech. Eng., Purdue Univ., 177 South Russel St., West Lafayette, IN 47907-2099, bntong@purdue.edu)

The Lorentz transform, which can be applied to a uniformly moving source, effectively converts the time-domain wave equation to an equivalent problem in frequency-domain due to a stationary source. Standard fast field program (FFP) implementations apply the property of conjugate symmetry in the wavenumber domain for improved efficiency. Recent literature [D. Dragna *et al.*, AIAA **52**, 1928–1939 (2014)] suggests the use of a Dopplerized frequency-dependent impedance model to account for the effects of source motion. Consequently, the FFP kernel function is no longer identical for the positive and negative wavenumbers. Additional complications are introduced by the necessity to compute the positive and negative horizontal separation distances in the Lorentz frame to obtain a complete time history of sound pressures for the source approaching and receding from the receiver. Further development of the FFP algorithm in the Lorentz frame is explored such that a frequency-dependent impedance model is developed. Both moving line and point monopole sources are considered in the current investigation. Results are validated against direct numerical integration schemes and with the published time-domain solutions. Applications for an aircraft operating in cruise condition is also examined in the present study. [Work Sponsored by the Federal Aviation Administration.]

Session 2pPP

Psychological and Physiological Acoustics, Biomedical Acoustics, Speech Communication, and Signal Processing in Acoustics: Celebration of the Modern Cochlear Implant and the First Substantial Restoration of a Human Sense Using a Medical Intervention II

Blake S. Wilson, Cochair
Duke University, 2410 Wrightwood Ave., Durham, NC 27705

Michael Dorman, Cochair
Department of Speech and Hearing Science, Arizona State University, Tempe, AZ 85258

Invited Papers

1:00

2pPP1. Lessons learned from multichannel cochlear implants that are relevant to a visual prosthesis. Jim Patrick (Cochlear Ltd., Macquarie Univ., 1 University Ave., New South Wales 2109, Australia, jpatrick@cochlear.com)

The clinical success of the multichannel cochlear implant was built on the successful outcomes of the multidisciplinary research program led by Professor Graeme Clark at the University of Melbourne and many of the outcomes from these studies can also be applied to other sensory prostheses, including the visual prosthesis. A primary requirement of any neural prosthesis is that it is safe, and this presentation will include studies that define stimulus parameters that allow stable long term responses, with no evidence of metabolic overload or neural degeneration. Any device that is chronically implanted for the life of a patient must be reliable, with a low likelihood of need for replacement because of device failure or infection, and the presentation will include both the results from Clark's infection studies and long term cochlear implant reliability observations. The presentation will describe the design of an experimental retinal prosthesis that uses an electrode array that is based on an array that was used to stimulate the cochlear nucleus of profoundly deaf patients. It will include preliminary results for volunteer patients, and approaches that may improve these pilot study outcomes.

1:20

2pPP2. Lessons learned from cochlear implants that are relevant to a vestibular prosthesis. Jay T. Rubinstein, Chris Phillips, Kaibao Nie, Leo Ling, and James O. Phillips (Otolaryngol., Univ. of Washington, Box 357923, Seattle, WA 98195, rubinj@uw.edu)

In October, 2010, we began human studies of a vestibular prosthesis. Our goal is to develop a device for the management of several vestibular disorders that defy existing treatment modalities. We have long-term data on four human subjects and ten non-human primates and are awaiting regulatory approval of our second-generation device, modified based on that data. We have been guided by a wealth of information based on lessons learned from cochlear implants. These lessons include first and foremost the otherwise hubristic idea that an effective vestibular prosthesis is technically feasible. Indeed the extraordinary compensation mechanisms of the central vestibular system convince us that such a prosthesis could be even more effective than for hearing. Second, the ability to leverage CI engineering expertise and industry relationships has been critical to our success so far. Third, the regulatory comfort with the success and favorable risk profile of cochlear implants has greatly speeded up human translation. Fourth, surgical familiarity with these devices has greatly enabled our research and facilitates future multicenter studies. Lastly, experience with "soft-surgery", hearing preservation, and single-sided deafness cochlear implantation holds forth alluring possibilities for the range of vestibular disorders that may eventually succumb to this closely related technology.

1:40

2pPP3. Neuroplasticity with cochlear implants: Underlying neuronal mechanisms. Andrej Kral (Dept. of Experimental Otology, Inst. of AudioNeuro Technol., Medical Univ. Hannover, Feodor-Lynen-Str. 35, Hannover 30625, Germany, a.kral@uke.de)

Chronic cochlear implant (CI) stimulation in congenitally deaf cats (CDCs) leads to cortical maturation matching imaging data from humans (Kral and Sharma, 2012, TINS). One possible measure of auditory plasticity is the reorganization of aural preference following monaural CIs (Kral *et al*., 2013, Brain), showing a sensitive period of <4.2 months in cats. A substantial reduction of binaural information following developmental unilateral hearing was found in cortical neurons (Kral *et al*., 2015, Audiol Neurootol). Consequently, auditory maturation requires experience. Additional to reduced synaptic plasticity, loss of acuity in feature representation, deficits in integrative function of the cortical column and deficits in corticocortical, particularly top-down, interactions, close the sensitive periods (Kral 2013, Neuroscience). Cross-modal plasticity recruits auditory resources for non-auditory tasks. Despite of cross-modal reorganization in some auditory areas the extend of the underlying reorganization of corticortical connections (Barone *et al*., 2013, PLoS One) indicates that this only moderately limits auditory processing capacity. Electrophysiological recordings in the dorsal auditory cortex of CDCs demonstrate that the cross-modally reorganized secondary auditory areas maintain a predominance of dormant auditory inputs additional to moderate cross-modal reorganization. [*Supported by Deutsche Forschungsgemeinschaft (Cluster of Excellence Hearing4all).]*

2:00

2pPP4. Longitudinal results in the childhood development after cochlear implantation study. John K. Niparko (Dept. of Otolaryngology-Head & Neck Surgery, Keck School of Medicine, Univ. of Southern California, Los Angeles, CA 90033, niparko@med.usc.edu)

This NIDCD-funded study has aimed to develop a longitudinal, multivariate model of spoken language outcomes after early cochlear implantation. Our national cohort accrued subjects in their infant and toddler stages. As the study enters its 12th year, the study offers a model of post-implant language development, now at a stage when participants are able to self-assess their interaction with peers, school performance, and quality of life. Here, we will present our primary outcome of interest-spoken language acquisition-discussing modifying variables related to auditory function, cognition, device, and environmental variables.

2:20

2pPP5. Importance of cochlear health for cochlear-implant function. Bryan E. Pfingst (Dept. of Otolaryngol., Univ. of Michigan, Kresge Hearing Res. Inst., 1150 West Medical Ctr. Dr., Ann Arbor, MI 48109-5616, bpfingst@umich.edu), Ning Zhou (Dept. of Commun. Sci. and Disord., East Carolina Univ., Greenville, NC), Deborah J. Colesa, Melissa M. Watts (Dept. of Otolaryngol., Univ. of Michigan, Ann Arbor, MI), Stefan B. Strahl (MED-EL GmbH, Innsbruck, Austria), Soha N. Garadat (Dept. of Hearing and Speech Sci., Univ. of Jordan, Amman, Jordan), Kara C. Schvartz-Leyzac, Yehoash Raphael, and Teresa A. Zwolan (Dept. of Otolaryngol., Univ. of Michigan, Ann Arbor, MI)

In humans with cochlear implants, functional measures show considerable variation from one stimulation site to another along the electrode array. Our research has demonstrated that (1) the across-site patterns of the functional data are stable over time but differ across subjects; and (2) the across-site patterns are measure specific. These observations are consistent with the hypotheses that implant performance at a given stimulation site is dependent on specific conditions near the site, and that the various functional measures do not all depend on the same conditions. However, we lack direct evidence as to the specific conditions leading to the site-specific differences in performance in humans. Studies in our guinea pig laboratory and elsewhere have demonstrated highly significant correlations between psychophysical or electrophysiological measures of implant function and anatomical measures of cochlear health. Furthermore, the correlated anatomical features differ across functional measures. Finally, some functional measures that are correlated with anatomical measures of cochlear health in animals are predictive of speech recognition ability in human implant users. The data support efforts to preserve and/or restore the health of the implanted cochlea. [This work was supported by NIH/NIDCD grants R01 DC010786, R01 DC010412, and P30 DC05188, and a contract from MED-EL.]

2:40–2:55 Break

2:55

2pPP6. Engineering contributions to the development of the cochlear implant. Fan-Gang Zeng (Otolaryngol. - Head and Neck Surgery, Univ. of California Irvine, 110 Med Sci E, Irvine, CA 92697, fzeng@uci.edu)

The cochlear implant is a product of deliberate systems engineering, starting with a clear design goal by physicians to restore human hearing by means of electrical stimulation. The engineering development of the cochlear implant has been a dynamic interplay between a perfectionist's approach attempting to replicate the intricate sensing and processing in a normal biological system and a reductionist's approach building an artificial system with minimal components and complexity. The reductionist's approach won the initial battle as the first FDA-approved single-channel cochlear implant consisted of simply a microphone, an amplifier, and a pair of coils. Borrowing techniques and knowhow's from diverse engineering areas in acoustics, aerospace, and electronics, modern multi-channel cochlear implants have shifted towards the perfectionist's approach. Acoustic researchers, particularly speech scientists, have played an important role in the engineering development of the cochlear implant. From an engineering perspective, I will highlight development milestones, evaluate their contributions to actual implant performance, and delineate the complicated relationships between artificial and natural hearing.

3:15

2pPP7. Anatomical considerations for the design and applications of cochlear implants. Helge E. Rask-Andersen, Wei Liu (Dept. ORL, Uppsala Univ. Hospital, Uppsala, Uppsala 75241, Sweden, helge.raskandersen@gmail.com), Annelies Schrott-Fischer, and Rudolf Glueckert (Dept. ORL, Medical Univ. of Innsbruck, Innsbruck, Austria)

This presentation aims to display some important anatomical characteristics and variations of the human cochlea that may influence outcomes with cochlear implants. I will describe the complexity of the "hook" region and how it may challenge atraumatic insertion of an electrode array. In addition, I will discuss advantages and disadvantages of various cochleostomy approaches and of various trajectories for insertions. These anatomical considerations are informed by micro dissections of human temporal bones and plastic casts of different parts of the bones. In addition, the studied microanatomy of the basilar membrane may inform insertions and insertion depths needed to preserve cochlear structures. In particular, the studies suggest that the vulnerability of the membrane may be greatest at the apex and this possibility has implications for electrode designs and insertions. Also, the apical region may be most susceptible to inflammation and fibrosis due to its anatomy. The anatomical considerations for safe insertions of electrodes, including preservation of the remaining neural tissue, will be discussed. This work was supported by grants from Uppsala University Hospital and Uppsala University; the Tysta Skolan Foundation; the Swedish Deafness Foundation; the European Community 7th Framework Programme; and kind private donations from Börje Runögård.

3:35

2pPP8. Tissue engineering and pharmaceutical interventions. Josef M. Miller, Yehoash Raphael, Keith R. Duncan, Richard A. Altschuler, and Gabriel Corfas (Kresge Hearing Res. Inst., Univ. of Michigan, Ann Arbor, MI 48109, josef@umich.edu)

Hearing performance with modern cochlear implants is dependent on remaining sensory and neural elements. CNS plasticity and individual capacity for outstanding performance with meager peripheral neural survival provide some exceptions; however, both human and animal studies support a direct relationship of performance with structural integrity/function of remaining sensory and neural elements in the implanted ear. This dependency will increase with next generation implants. This has driven interventions based on increased understanding of molecular mechanisms underlying cell death and survival to enhance sensory and neural survival. It has driven tissue engineering of remaining elements with neurotrophic and other factors to induce regeneration, neurite outgrowth, and stem cell replacements of lost elements. Delivery to the cochlea is being accomplished via middle ear or transtympanic application, direct delivery in bolus or continuous administration via osmotic pumps, as coatings or channels in implants, gene transfer via viral vectors, factors incorporated in nano- and micro-particles, and via systemic administration. Current results provide a base for optimism for a future integrated system of cochlear implant-drug/gene delivery system. Each agent and delivery has risks and benefits. Challenges to select the best strategy for translation and application in humans will be discussed. [This work was s supported by NIH/NIDCD grants R01 DC010412, DC 004820, DC011294, P30 DC005188, and DOD W81XWH-12-10492.]

TUESDAY AFTERNOON, 19 MAY 2015 KINGS 3, 1:30 P.M. TO 2:45 P.M.

Session 2pSAa

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

Christina J. Naify, Cochair
Acoustics, Naval Research Lab, 4555 Overlook Ave. SW, Washington, DC 20375

Michael R. Haberman, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Contributed Papers

1:30

2pSAa1. A transformation-based formulation of an airfoil cloak using Joukowski mapping. Saliou Telly (Mech. Eng., Univ. of Maryland College Park, 14359 Long Channel Dr., Germantown, MD 20874, stelly@umd.edu) and Balakumar Balachandran (Mech. Eng., Univ. of Maryland College Park, College Park, MD)

Since its introduction in 2006, the transformation approach to cloaking has been extensively explored for electromagnetic and acoustic fields, due to its intuitive nature as well as the methodology. In this approach, the construction of invisibility cloaks consists of exploiting a change of coordinates to create a void region within a finite space while keeping the outer boundary of the considered space unchanged. Such changes in physical coordinates can be interpreted as a transformation of material properties related to electromagnetics and acoustics, given the invariance of the respective field equations to coordinate change. Based on the transformation, properties of an invisibility cloak can be inferred using available formulae. To this date, this approach has been successfully applied to various two-dimensional geometries including circular and square cylinders. In this work, as an extension, the authors present initial results obtained for the cloaking of a two-dimensional airfoil section. They take advantage of the mapping properties of the well-known Joukowski transformation, a complex mapping that is used in aerodynamics applications.

1:45

2pSAa2. An impedance-mobility model of stacked membrane-type acoustic metamaterials. Matthew G. Blevins (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 909 S 70th plz #4, Omaha, NE 68106, mblevins@huskers.unl.edu), Siu-Kit Lau (Armstrong (China) Investment Co. Ltd., Shanghai, China), and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, Omaha, NE)

Membrane-type acoustic metamaterials have been proven to exhibit high low-frequency transmission loss despite their small thickness and light weight. To date, analysis has focused primarily on experimental studies in plane-wave tubes and numerical modeling using finite element methods. These methods are inefficient when used for applications that require iterative changes to the structure of the material. In addition, high sound transmission loss with a single layer of such metamaterial only occurs in a narrow frequency range. To facilitate design and optimization of stacked membrane-type acoustic metamaterials, a computationally efficient dynamic model based on the impedance-mobility approach is proposed. Results are verified against a finite element model. Single and double layer transmission loss characteristics are compared. Wide-band high-transmission-loss acoustic metamaterials can be achieved by double layer membranes and using the proposed approach for optimization. The impedance-mobility approach is shown to be very efficient for modeling and optimization of such materials, compared against the conventional finite element approach.

2:00

2pSAa3. Acoustoelectric admittance of two-dimensional ferroelectric metamaterial under dispersion curve branching. Ola H. Nusierat (Dept. of Phys. and Astronomy, Univ. of MS, 108 Lewis Hall, P.O. Box 1848, University, MS 38677, ohnusier@go.olemiss.edu), Lucien Cremaldi, and Igor Ostrovskii (Phys. and Astronomy, Univ. of MS, Oxford, MS)

The acoustoelectric admittance Y of the ZX-cut periodically poled $LiNbO_3$ plate is investigated experimentally and theoretically computed. The sample consists of 88 inversely poled domains of 0.45 mm-long each. The vector voltmeter, digital oscilloscope, and function generator were used to measure the frequency dependencies Y(F) at room temperature. Double peaks were observed in the admittance measurements near the lower edge of a stop-band at frequencies 3.268 MHz and 3.287 MHz. Another double peak exists near upper edge of the acoustic stop band at frequencies 3.651 MHz and 3.663 MHz. The double peak in Y can be explained as follows. Ultrasound in this ferroelectric acoustic metamaterial (FAM) has a so-called stop band when an acoustic wavelength is close to a double-length of ferroelectric domain within the inversely poled structure. The dispersion curves computed by the finite element method reveal an effect of decoupling of two acoustic displacements in a zero-antisymmetric mode. The two displacements Ax and Az along the X and Z axes become decoupled near the boundaries of the acoustic Brillouin zone. This can be explained by different diffraction losses in two orthogonal displacements within the FAM. Computations are in a good agreement with experiments.

2:15

2pSAa4. Acoustic Poisson-like effect. Alexey Titovich and Andrew N. Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, alexey.titovich@rutgers.edu)

A sonic crystal with a particular local resonance is capable of transferring incident acoustic energy to a perpendicular direction. This is done with a square array of cylindrical scatterers vibrating with a non-axisymmetric mode which couples to the anti-symmetric mode (deaf mode) of the crystal via an evanescent band unlocking it from the bandgap boundaries and resulting in perpendicular propagation. The scatterer is an elastic shell designed to be acoustically transparent with impedance and index matched to water, which is vibrating with the $n=2$ in-plane bending mode. The shells' collective motion resembles the quasi-static Poisson effect in elastic solids and results in a quadrupole scattering pattern which causes a purely perpendicular mode to be excited at normal incidence. Thus, the acoustic Poisson-like effect is non-refractive and also highly efficient. Variations of this novel effect will also be discussed such as different local resonances, lattices and crystals. [Support from ONR is gratefully acknowledged.]

2:30

2pSAa5. Acoustic ground cloaks revisited. Peter Kerrian (Graduate Program in Acoust., The Pennsylvania State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, pak215@psu.edu), Amanda Hanford, Dean Capone, and Scott Miller (Appl. Res. Lab Penn State, University Park, PA)

The unique material properties now obtainable with acoustic metamaterials have led to unprecedented control of acoustic wave propagation, resulting in many applications including acoustic cloaking. The two fundamental approaches in the development of a ground cloak are quasiconformal mapping [Li *et al.* Phys. Rev. Lett. **101**, 203901 (2008)] and coordinate transformations [Popa *et al.* Phys. Rev. B. **83**, 224304 (2011)]. The differences in the required material properties prescribed by these two approaches lie in the amount of anisotropy and inhomogeneity, as well as the size of the cloak relative to object. The coordinate transformation approach has been used to produce a realizable anisotropic homogeneous ground cloak in the acoustic domain. This presentation will highlight the findings of work that examined how advances in metamaterial development could lead to the realization of required material properties for ground cloaks, and explore alternative transformations to expand the applications for acoustic ground cloaks.

TUESDAY AFTERNOON, 19 MAY 2015 COMMONWEALTH 2, 3:30 P.M. TO 4:50 P.M.

Session 2pSAb

Structural Acoustics and Vibration and Signal Processing in Acoustics: Time-Domain Methods

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

John B. Fahnline, Cochair
ARL / Penn State, P.O. Box 30, State College, PA 16804-0030

Invited Papers

3:30

2pSAb1. Inverse problems in transient structural acoustics. Timothy F. Walsh (Computational Solid Mech. and Structural Dynam., Sandia National Labs., PO Box 5800, MS 0380, Albuquerque, NM 87185, tfwalsh@sandia.gov) and Wilkins Aquino (Civil and Environ. Eng., Duke Univ., Durham, NC)

Inverse problems are frequently encountered in transient structural acoustics. Given measured accelerometer or microphone time histories, one may need to estimate material properties in a structure, impedance conditions on the boundaries, shape or topology of an object, or characterize acoustic and structural sources that produced the measured data. Applications include model calibration, acoustic testing of aerospace structures, military surveillance, and underwater acoustics. Typically, accelerometer or microphone pressures are

measured experimentally, and it is desired to characterize the material parameters, boundary conditions, topology, or acoustic sources that produced these accelerations or microphone pressures. In this talk, we will present a set of example inverse problems in structural acoustics, and a framework for their solution using an operator-based partial differential equation (PDE) constrained optimization approach. This abstract framework enables one to reduce any inverse problem to a set of fundamental operations and enables re-use of several software constructs. Formulations will be presented in both the time and frequency domain, and the merits and drawbacks of both approaches will be discussed. [Sandia National Laboratories is a multi-program laboratory managed and operated by Sandia Corporation, a wholly owned subsidiary of Lockheed martin Corporation, for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-AC04-94AL850000.]

3:50

2pSAb2. Simulation of coupled structural-acoustic response with dynamic damage evolution. Jonathan S. Pitt (Appl. Res. Lab., The Penn State Univ., Appl. Res. Lab, PO Box 30, Mailstop 3320B, State College, PA 16804, jonathan.pitt@psu.edu)

A novel time-domain method for simulating dynamic damage evolution in a coupled structural-acoustic system is presented. The system is derived via the theory of continuum damage mechanics, and incorporates standard damage evolution models. The overall solution method is staggered, solving for the dynamic damage evolution first with an explicit step, and then using the new values in the coupled computation of the structural-acoustic system. The spatial domain is discretized using a mixed finite element method, and the temporal space is discretized with a higher-order implicit time discretization scheme. Efforts toward fully coupled verification of the solution algorithm are presented, as are validation studies for cases without evolving damage. Applications with evolving damage are presented, and present a first principles study of changes in the structural acoustic response to dynamically evolving damage in the structure. Examples of downstream usage of the evolving structural response are discussed in the concluding remarks.

4:10

2pSAb3. Time windowed comparisons between models and measurements. James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jgm@bu.edu)

The present work proposes and demonstrates a method for improving the accuracy of large finite element models using measurements. Finite element analysis routinely allows for the construction of dynamic models with millions of degrees of freedom. The models often involve a large number of parameters, such as material properties and dimensions, that are not precisely known. For example, one may measure the material properties of a homogeneous sample but those properties will vary from sample to sample. Moreover, material properties may change considerably during construction as the material is prestressed, bonded, or welded. Therefore, it is often the case that the finite element model does not agree with measurements and one is faced with the task of modifying model parameters to yield agreement. This is made difficult by the large number of parameters as well as the computational cost of evaluating the model for each set of parameter choices. The present work proposes a time windowing method that localizes the spatial volume of response in the model and experiment and therefore isolates a small number of model parameters that may be varied to bring agreement. Numerical examples are presented to illustrate the method.

4:30

2pSAb4. A coupled finite element/equivalent source formulation for transient structural-acoustic problems. John B. Fahnline (ARL / Penn State, P.O. Box 30, State College, PA 16804-0030, jbf103@arl.psu.edu)

Recent research suggests that instability problems that occur in the time domain boundary element formulations are manifestations of nonuniqueness difficulties, and that remedies similar to those used in frequency domain formulations can address the difficulties and improve solution stability. For frequency domain equivalent source formulations, the difficulties can be addressed by combining simple and dipole sources together to form "tripole sources". The basic goals of the current research are to develop an analogous equivalent source formulation for transient acoustic boundary value problems and to combine it with structural finite element analyses to solve transient coupled structural-acoustic problems. A brief description of the equivalent source formulation is given along with several validation cases for acoustic boundary value problems, including one with a closed boundary surface where nonexistence difficulties should occur. The formulation requires convolution summations, which are independent of each other and a brief discussion is given of how the computations can be parallelized. A brief description is also given of the coupled finite element/equivalent source formulation for transient structural-acoustic boundary value problems. Several examples are given to validate the formulation and to demonstrate that the tripole source formulation for the equivalent sources is more stable than the simple source version.

Session 2pSC

Speech Communication: Speech Methods, Models and Technology (Poster Session)

Robert Hagiwara, Chair
Department of Linguistics, University of Manitoba, Winnipeg, MB R3T 5V5, Canada

Posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to view other posters, contributors of odd-numbered papers will be at their posters from 1:15 p.m. to 2:45 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 4:30 p.m. There will be a 15-minute break from 2:45 p.m. to 3:00 p.m.

Contributed Papers

2pSC1. Channel and noise robustness of articulatory features in a deep neural net based speech recognition system. Vikramjit Mitra (Speech Technol. and Res. Lab, SRI Int.., 333 Ravenswood Ave., EJ133, Menlo Park, CA 94025, vmitra@speech.sri.com), Ganesh Sivaraman (Univ. of Maryland, College Park, MD), Hosung Nam (Haskins Labs, New Haven, CT), Carol Y. Espy-Wilson (Univ. of Maryland, College Park, MD), and Elliot Saltzman (Boston Univ., Boston, MA)

Articulatory features (AFs) are known to provide an invariant representation of speech, which is expected to be robust against channel and noise degradations. This work presents a deep neural network (DNN)—hidden Markov model (HMM) based acoustic model where articulatory features are used in addition to mel-frequency cepstral coefficients (MFCC) for the Aurora-4 speech recognition task. AFs were generated using a DNN trained layer-by-layer using synthetic speech data. Comparison between baseline mel-filter-bank energy (MFB) features, MFCCs and fusion of articulatory feature with MFCCs show that articulatory features helped to increase the noise and channel robustness of the DNN-HMM acoustic model, indicating that articulatory representation does provide an invariant representation of speech.

2pSC2. A method for estimating lingual cavity volume in click consonants. Amanda L. Miller (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210-1298, miller.5592@osu.edu)

A method for estimating lingual cavity volume in click consonants is presented. The upper edge of the lingual cavity is estimated from stone palate casts, which were scanned using a Polhemus 3D Digitizer. The floor of the midline of the lingual cavity was estimated using 114 fps mid-sagittal ultrasound traces of the tongue collected with the CHAUSA method (Miller and Finch 2011). The width of the cavity from front to back was estimated by measurements of the cavity width at six different locations in linguograms. Changes in lingual cavity volume for the four coronal click types in Mangetti Dune !Xung are estimated by changing the mid-sagitttal ultrasound trace that corresponds to one of five stages in click production. 3-D images of the changes in cavity volume for each of the four coronal click types recognized by the IPA (2006) are presented.

2pSC3. Quantal biomechanics in an embodied phonetics. Bryan Gick (Linguist, Univ. of Br. Columbia, 2613 West Mall, Vancouver, British Columbia V6T1Z4, Canada, gick@mail.ubc.ca) and Scott R. Moisik (The Max Planck Inst. for PsychoLinguist, Nijmegen, Netherlands)

Quantal regions were described by Stevens [e.g., 1989, J. Phon. **17**, 3–45] to identify nonlinear stabilities in the relationship between articulation and acoustics. Classic cases of quantal effects show how tongue posture may vary within one region of the vocal tract with little acoustic change, while in other regions very small movements can have large effects on acoustic output. Such effects can be thought of as attractors to speech behavior in those regions of the phonetic space that allow greater noise. Quantal-like stabilities have been suggested to operate not just in articulatory-acoustic space, but in biomechanical-articulatory space as well [e.g., Schwartz *et al*., 1997, J. Phon. **25**, 255–286]. It is argued here that such quantal-like stabilities are a hallmark of speech modules [Gick & Stavness, 2013, Front. Psych. **4**, 977], providing the basis for robust, feed-forward control. Computer simulations in the ArtiSynth platform (www.artisynth.org) are used to demonstrate quantal effects in speech biomechanics at multiple vocal tract loci, including the lips, oropharyngeal isthmus, and larynx. Moving into the future, quantal work will integrate observations about nonlinear stabilities cutting across the many physical and sensory domains that figure in speech. [Research funded by NSERC.]

2pSC4. Measurement of child speech complexity using acoustic landmark detection. Marisha Speights, Suzanne Boyce (Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., Cincinnati, OH 45267, speighma@mail.uc.edu), Joel MacAuslan (Speech Technol. and Appl. Res., Boston, MA), and Harriet Fell (College of Comput. Information Sci., Northeastern Univ., Boston, MA)

An important measure of intelligibility in children is the ability to articulate complex syllables. At the same time, the transcription and analysis of young children's speech data is laborious and time-intensive, requiring specialized training and expert perceptual abilities on the part of researchers. Researchers have called for automatized methods to quantify these measures of articulatory complexity. One such method, Automatic Syllabic Cluster Analysis, may allow for more efficient analysis by using acoustic landmarks [Stevens *et al*.,1992] to systematically detect abrupt and maximal events in the speech signal, group them into syllable patterns and further group the syllable patterns into utterances. Statistics derived from these groupings are used to determine the complexity of utterances. Automatic Syllabic Cluster Analysis does not require transcription for analysis of the speech because it is not lexically driven, instead employing measurements of acoustic parameters that characterize changes in articulatory precision. To identify a potential application for this automated approach, this pilot research examines syllabic complexity in children to differentiate between children who are typically developing and those with diagnosed speech disorders. Preliminary results indicate that Automatic Syllabic Cluster Analysis can identify articulatory differences between the groups.

2pSC5. Augmenting acoustic phonetics with articulatory features for phone recognition. Ganesh Sivaraman (Elec. & Comput. Eng., Univ. of Maryland College Park, 5002 Seminole St., Berwyn Heights, MD 20740, ganesa90@umd.edu), Vikramjit Mitra (SRI Int.., Menlo Park, CA), Hosung Nam (Korea Univ., Seoul, South Korea), Elliot Saltzman (Dept. of Physical Therapy and Athletic Training, Boston Univ., Boston, MA), and Carol Espy-Wilson (Elec. & Comput. Eng., Univ. of Maryland College Park, College Park, MD)

In articulatory phonetics, a phoneme's identity is specified by its articulator-free (manner) and articulator-bound (place) features. Previous studies have shown that acoustic-phonetic features (APs) can be used to segment speech into broad classes determined by the manner of articulation of speech sounds; compared to MFCCs, however, APs perform poorly in determining place of articulation. This study explores the combination of APs with vocal Tract constriction Variables (TVs) to distinguish phonemes according to their place of articulation for stops, fricatives and nasals. TVs were estimated from acoustics using speech inversion systems trained on the XRMB database with pellet trajectories converted into TVs. TIMIT corpus sentences were first segmented into broad classes using a landmark based broad class segmentation algorithm. Each stop, fricative and nasal speech segment was further classified according to its place of articulation: stops were classified as bilabial (/P/, /B/), alveolar (/T/, /D/) or velar (/K/, /G/); fricatives were classified as labiodental (/F/, /V/), alveolar (/TH/, /DH/, /S/, /Z/), palatal (/SH/, /ZH/) or glottal (/HH/); and nasals were classified as bilabial (/M/), alveolar (/N/) or velar (/NG/). Polynomial kernel support vector machines were trained on APs concatenated with vocal tract constriction features. Results showed that combining acoustic and articulatory features leads to reliable recognition of manner and place of articulation, and improves phone recognition.

2pSC6. Non-distorting method of vowel normalization by transposition in the semitonal scale. Danila Gomulkin (Povarskoy per. 3, kv. 26, Saint Petersburg 191025, Russian Federation, gomulkin@yahoo.co.uk)

Converting formant values from Hertz into semitones provides a remarkable "area agreement" between vowel systems in the F1/F2 plane without distorting intervals between the formants. Further simple transposition of a Sample system in relation to a Reference system by the difference between the mean of all converted F_1s and F_2s for the Sample speaker/group and the mean of those for the Reference speaker/group efficiently eliminates physiological mismatch between the speakers/groups and allows for convenient comparison of the vowel systems in a meaningful scale of pitch intervals. To convert formant *n* of vowel *V* from Hertz to semitones (expressed in MIDI keyboard numbers) use formula: [1] $F_{n[V]midi} = \text{midi}(F_{n[V]Hz})$, where [2] $\text{midi}(x) = 12{*}\log2(x/440) + 69$. To normalize converted formants of a Sample speaker/group against those of a Reference speaker/group, deduct from the converted Sample formants ($F_{n[V]midi\ (Sample)}$) the difference between the mean of all converted F_1s and F_2s for the Sample speaker/group and the mean of those for the Reference speaker/group (Δ $\text{MEAN}_{midi(Sample-Ref)}$): [3] $F^*_{n[V]midi\ (Sample)} = F_{n[V]midi\ (Sample)} - \Delta\ \text{MEAN}_{midi(Sample-Ref)}$

2pSC7. Realtime voice activity and pitch modulation for laryngectomy transducers using head and facial gestures. Gautam Mohan, Katherine Hamilton, Andrew Grasberger, Adam C. Lammert, and Jason Waterman (Comput. Sci. Dept., Swarthmore College, 500 College Ave., Swarthmore, PA 19081, lammert@cs.swarthmore.edu)

Individuals who have undergone laryngectomy often rely on handheld transducers (i.e., the electrolarynx) to excite the vocal tract and produce speech. Widely used electrolarynx designs are limited, in that they require manual control of voice activity and pitch modulation. It would be advantageous to have an interface that requires less training, perhaps using the remaining, intact speech production system as a scaffold. Strong evidence exists that aspects of head motion and facial gestures are highly correlated with gestures of voicing and pitch. Therefore, the goal of project MANATEE is to develop an electrolarynx control interface which takes advantage of those correlations. The focus of the current study is to determine the feasibility of using head and facial features to accurately and efficiently modulate the pitch of speaker's electrolarynx in real time on a mobile platform using the built-in video camera. A prototype interface, capable of running on desktop machines and compatible Android devices, is implemented using OpenCV for video feature extraction and statistical prediction of the electrolarynx control signal. Initial performance evaluation is promising, showing pitch prediction accuracies at double the chance-level baseline, and prediction delays well below the perceptually-relevant, ~50 ms threshold.

2pSC8. Dual electromagnetic articulometer observation of head movements coordinated with articulatory gestures for interacting talkers in synchronized speech tasks. Mark Tiede (Haskins Labs., 300 George St., Ste. 900, New Haven, CT 06511, tiede@haskins.yale.edu) and Dolly Goldenberg (Linguist, Yale Univ., New Haven, CT)

Previous research has demonstrated that speakers readily entrain to one another in synchronized speech tasks (e.g., Cummins 2002, Vatikiotis-Bateson *et al.* 2014, Natif *et al.* 2014), but the mixture of auditory and visual cues they use to achieve such alignment remains unclear. In this work, we extend the dual-EMA paradigm of Tiede *et al.* (2012) to observe the speech and coordinated head movements of speaker pairs interacting face-to-face during synchronized production in three experimental tasks: the "Grandfather" passage, repetition of short rhythmically consistent sentences, and competing alternating word pairs (e.g., "topper-cop" vs. "copper-top"). The first task was read with no eye contact, the second was read and then produced with eye contact, and the third required continuous eye contact. Head movement was characterized using the tracked position of the upper incisor reference sensor. Prosodic prominence was identified using F0 and amplitude contours from the acoustics, and gestural stiffness on articulator trajectories. Preliminary results show that frequency and amplitude of synchronized head movement increased with task eye contact, and that this was coordinated systematically with both acoustic and articulatory prosodic prominence. [Work supported by NIH.]

2pSC9. Benefits of using polar coordinates for working with ultrasound midsagittal tongue contours. Matthias Heyne (Linguist, Univ. of Canterbury, Private Bag 4800, Christchurch, Canterbury 8140, New Zealand, matthias.heyne@pg.canterbury.ac.nz) and Donald Derrick (New Zealand Inst. of Lang. Brain and Behaviour, Univ. of Canterbury, Christchurch, New Zealand)

Calculating SSANOVA average curves of tongue contours is a technique widely used in the field of speech research to facilitate the comparison of articulatory data (Davidson, 2006). Even though this technique is visually easier to understand and offers better statistical tests than using a concentric grid, Mielke (JASA, in press) has recently shown problems arising from generating SSANOVA curves in the Cartesian plane and has instead suggested the use of polar coordinates. Transferring Cartesian coordinates to the polar plane requires choosing an origin for the polar coordinate system and we hereby propose an alternative way of doing so by estimating the ultrasound transducer position through plotting two lines on an ultrasound image. The resulting average curves are very similar to Mielke's results but facilitate further analysis of data due to being relative to the estimated transducer position. Such possibilities include cutting off individual tokens at the edges to avoid wiggly ends of average contours, rotating (by adding/subtracting to/from the theta coordinate) and shifting data (by changing the x- and y-coordinates of the estimated origin in the Cartesian plane) to improve the (visual) comparability of ultrasound data across subjects. Some of these techniques were used in Heyne and Derrick (SST, 2014).

2pSC10. On the use of respiratory masks for speech measurements. Hao Zhang, Xiao Chen, and Stephen Zahorian (Dept. of Elec. and Comput. Eng., State Univ. of New York at Binghamton, Binghamton University, Binghamton, NY 13902, hzhang20@binghamton.edu)

Ideally, respiratory masks, when used to make certain measurements of speech and singing, should not interfere with what is being measured. Unfortunately this is not always the case. In this paper, two masks intended for speech measurements, are experimentally compared. One is a hard-walled mask manufactured and marketed by Kay-Pentax. The other is the circumferentially-vented pneumotachograph (CV) mask manufactured and marketed by Glottal (Syracuse, NY). Distortion measures, as quantified by formant frequencies, and muffling measures, as quantified by estimated mask transfer functions, indicate that the hard-walled mask interferes with the speech characteristics much more than does the CV mask. It is hypothesized that SPL measurements are also likely to be inaccurate if taken when using the hard-walled mask.

2pSC11. Audio dilation in real time speech communication. John S. Novak (Dept. of Comput. Sci., Eng. Res. Facility (ERF), Univ. of Illinois at Chicago, 842 W Taylor, Rm. 2032, M/C 152, Chicago, IL 60607, john.novak@gmail.com), Jason Archer (Dept. of Commun., Univ. of Illinois at Chicago, Chicago, IL), Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ., Chicago, IL), and Robert V. Kenyon (Dept. of Comput. Sci., Univ. of Illinois at Chicago, Chicago, IL)

Algorithmically decreasing speech tempo, or "audio dilation," can improve speech perception in attentionally demanding tasks [Gygi & Shafiro (2014), Hear. Res. **310**, pp. 76–86]. On-line audio dilation [Novak, *et al.* (2013), Interspeech 1869–1871] is a recently developed technique that decreases the tempo of audio signals as they are generated. This pilot study investigated effects of on-line audio dilation on performance in interactive problem solving tasks. We used a Diapix task [Baker & Hazan (2011), Behav. Res. Methods **43**(3), 761–770] to elicit and record spontaneous speech from pairs of participants under various dilation conditions: participants, seated in different rooms, were asked to find ten differences on two similar pictures, while their speech was either transmitted as spoken or dilated. Conditions tested include stretching one, both, or neither audio signal by 40%. Subsequent analysis shows that the technique, even using this substantial increase, did not interfere with interactive problem solving tasks, and did not lead to changes in speech production rate, measured as number of syllables per second. The lack of negative effects of on-line speech dilation provides a preliminary basis for further assessment of this method in speech perception tasks with high attentional and memory load.

2pSC12. Human-system turn taking analysis for the let's go bus information system. Tiancheng Zhao and Maxine Eskenazi (Lang. Technol. Inst., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, tianchez@andrew.cmu.edu)

We examined turn-taking in the Let's Go Bus spoken dialog data from two aspects: study the consequences of system barge-in when users are not finished speaking (false system barge-in, FSB); determine whether using a partial recognition result other than the final one produces better results. The consequence of FSBs is a less user-friendly system coupled with poor recognition caused by barge-ins, which divide one user turn into several fragments (UFs). We observed that UFs result in longer dialogs because the dialog manager has to recover from misrecognized utterances. Dialogs with UFs have 34 turns on average those without have 27. Poor recognition and long dialogs together cause lower task success rate. Dialogs with UFs have a success rate of 62% versus 84% for dialogs without. Moreover, we annotated the number of correct and incorrect slots for all partial recognitions. For 51% of the utterances, there exists a partial that contains more correct slots than the final recognition result. This will lead us to develop an algorithm to find the best partial. We conclude that systems that avoid FSB will have more efficient dialogs. They will also have better recognition by using the best partial instead of only the final one.

2pSC13. Robust automatic speech recognition in reverberation: onset enhancement versus binaural source separation. Hyung-Min Park (Dept. of Electron. Eng., Sogang Univ., Seoul, South Korea), Matthew Maciejewski (Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD), Chanwoo Kim (Google, Mountain View, CA), and Richard M. Stern (Dept. of Elec. and Comput. Eng., Carnegie Mellon Univ., Pittsburgh, PA, rms@cs.cmu.edu)

The precedence effect describes the auditory system's ability to suppress later-arriving components of sound in a reverberant environment, maintaining the perceived arrival azimuth of a sound in the direction of the actual source, even though the later reverberant components may arrive from other directions. It is also widely believed that precedence-like processing can also improve speech intelligibility for humans and the accuracy of speech recognition systems in reverberant environments. While the mechanisms underlying the precedence effect have traditionally been assumed to be binaural in nature, it is also possible that the suppression of later-arriving components may take place monaurally, and that the suppression of the corresponding components of the spatial image may be a consequence of this more peripheral processing. This paper compares potential contributions of onset enhancement (and consequent steady-state suppression) of the envelopes of subband components of speech at the monaural and binaural levels. Experimental results indicate that substantial improvement in recognition accuracy can be obtained in reverberant environments if feature extraction includes both onset enhancement and binaural interaction. Recognition accuracy appears to be relatively unaffected by which stage in the binaural processing is the site of the suppression mechanism. [Work supported by the LG Yonam Foundation and Cisco.]

2pSC14. Improving the accuracy of speech emotion recognition using acoustic landmarks and Teager energy operator features. Reza Asadi (College of Comput. and Information Sci., Northeastern Univ., 93 Chester St., Apt. 4, Allston, MA 02134, asadi@ccs.neu.edu) and Harriet Fell (College of Comput. and Information Sci., Northeastern Univ., Boston, MA)

Affective computing can help us achieve more intelligent user interfaces by adding the ability to recognize users' emotions. Human speech contains information about the emotional state of the speaker and can be used in emotion recognition systems. In this paper, we present a machine learning approach using acoustic features which improves the accuracy of speech emotion recognition. We used 698 speech samples from "Emotional Prosody Speech and transcripts" corpus to train and test the classifiers. The emotions used were happy, sadness, hot anger, panic, and neutral. Mel-frequency Cepstral Coefficients (MFCC), Teager Energy Operator (TEO) features, and acoustic landmark features were extracted from speech samples. Models were trained using multinomial logistic regression, k-Nearest Neighbors(k-NN) and Support Vector Machine(SVM) classifiers. The results show that adding landmark and TEO features to MFCC features improves the accuracy of classification. SVM classifiers with a Gaussian kernel had the best performance with an average accuracy of 90.43%. We achieved significant improvement in the accuracy of the classification compared to a previous study using the same dataset.

2pSC15. Using the speech recognition virtual kitchen infrastructure for reproducible cross-disciplinary speech research exchange. Andrew R. Plummer (Ohio State Univ., 2015 Neil Ave., Columbus, OH 43210, plummer.321@osu.edu)

Computational models and methods of analysis have become a mainstay in speech research over the last seventy years, but the means for sharing software systems is often left to personal communication between model developers. As a result, the sharing of systems is typically complicated, error-prone, or simply not done at all, making it difficult (in some cases impossible) to verify model performance, engage with developers directly using their own models, or bridge gaps between research communities that have diverged over time. Moreover, the learning curve for new students or tech consumers entering these communities is quite steep, limiting the use of the models to those initiated in a given area. Over the last few years a number of computing infrastructures have taken shape that aim to address the difficulties encountered in the exchange of large software systems (e.g., the Berkeley Computational Environment, http://collaboratool.berkeley.edu/). We present the Speech Recognition Virtual Kitchen (www.speechkitchen.org)—a computing infrastructure that provides tools and communication channels for the exchange of speech-related software systems in a manner that facilitates the sharing of reproducible research models and pedagogical material—together with several demonstrations of the infrastructure and its potential uses.

2pSC16. Documenting sound change with smartphone apps. Adrian Leemann (Dept. of Theor. and Appl. Linguist, Univ. of Cambridge, Sidgwick Ave., Cambridge, Cambridgeshire CB3 9DA, United Kingdom, al764@cam.ac.uk), Marie-José Kolly (LIMSI, Université de Paris-Sud, Paris, France), David Britain (Dept. of English Lang. and Linguist, Univ. of Bern, Bern, Switzerland), Ross Purves (Dept. of Geography, Univ. of Zurich, Zürich, Switzerland), and Elvira Glaser (Dept. of German, Univ. of Zurich, Zurich, Switzerland)

Crowdsourcing linguistic phenomena with smartphone applications is relatively new. Apps have been used to train acoustic models for automatic speech recognition (de Vries *et al.* 2014) and to archive endangered languages (Iwaidja Inyaman Team 2012). Leemann and Kolly (2013) developed a free app for iOS—*Dialäkt Äpp* (DÄ) (>78k downloads)—to document language change in Swiss German. Here, we present results of sound change based on DÄ data. DÄ predicts the users' dialects: for 16 variables, users select their dialectal variant. DÄ then tells users which dialect they speak. Underlying this prediction are maps from the *Linguistic Atlas of German-speaking Switzerland* (SDS, 1962-2003), which documents the linguistic situation around 1950. If predicted wrongly, users indicate their actual dialect. With this information, the 16 variables can be assessed for language change. Results revealed robustness of phonetic variables; lexical and morphological variables were more prone to change. Phonetic variables like *to lift* (variants: /lupfə, lʏpfə, lipfə/) revealed SDS agreement scores of nearly 85%, i.e., little sound change. Not all phonetic variables are equally robust: *ladle* (variants: /xælə, xællə, xæuə, xæɫə, xæɫɫə/) exhibited significant sound change. We will illustrate the results using maps that show details of the sound changes at hand.

2pSC17. The recorder's paradox: Balancing high-quality recordings with spontaneous speech in noisy recording environments. Paul De Decker (Linguist, Memorial Univ. of NF, Sci. Bldg. 3050B, St. John's, Newfoundland A1B 3X9, Canada, pauldd@mun.ca)

While sociophonetic analysis requires high-quality sound recordings, sociolinguistic interviews (Labov 1984) are often conducted in uncontrolled, natural environments to elicit casual speech (Tagliamonte 2006). The effects of room acoustics and background noise on formant measurements, however, have never been systematically examined. To empirically investigate how ambient noise affects recording quality and measurements, a male speaker of English was simultaneously recorded to multiple devices reading 260 carrier phrases. Three naturally occurring background noise conditions (+20dB, +10dB, and +0dB SNR) were created in Praat 5.4 (Boersma and Weenink 2014) and mixed with the original audio recordings. 10,028 measurements of F1 and F2 were taken at the temporal midpoint of each vowel in each of the four recordings using LPC analysis in Praat. Pearson's r tests in R (R Core Team, 2014) assessed the correlation between measurements from each recording. Main results reveal positive correlations between "noiseless" and +20dB, and +10dB SNR conditions for each device, all vowels and both formants. When the signal was not appreciably louder than the background noise (i.e. +0 dB SNR) correlations significantly weakened. These findings are discussed as they relate to sociolinguistic investigations that need to balance high-fidelity recordings with noisier speaking environments.

2pSC18. A coder with adaptive filters and non-uniform techniques for simple speech communication. Seonggeon Bae, Myungsook Kim, and Myungjin Bae (Soongsil Univ., 1118-408, Mokdong Apt. 11dangi, Yangcheon Gu, Seoul 148-771, South Korea, sgbae123@gmail.com)

This paper proposes a new method of non-uniform sampling coding that utilizes a modified zero-crossing rate filter and a variable filter. In speech signals, the modified zero-crossing rate refers to the number of times in which the signal crosses zero in a single frame; the rate is high in noisy sections but low in sections of vocal sounds that are quasi-periodic. And the modified zero-crossing rates of unvoiced signals are similar to those of voiced signals. In speech signals, the non-uniform sampling coding method utilizes the peaks and valleys in a frame. In noisy sections, there are a lot of zero crossings and thus creating many peaks and valleys, while sections with voice sounds have zero-crossings numbering only 2–3 times of their fundamental frequency. The method proposed in this paper utilizes the number of peaks and valleys in determining the validity of a sample; the less we have peaks and valleys, the higher compression rates we get. Therefore, by decreasing the number of samples from sections with noise and voiceless sounds which have a high number of peaks and valleys, and by creating a signal that has little effect on the recognized signals, we may acquire a better compression rate. As examined and proposed in this paper, these features can be used to design a filter coefficient for a better performance in order to improve compression rates of non-uniform sampling coding and to maintain high quality signals.

2pSC19. Vowel nasalization might affect the envelop of the vowel signal by reducing the magnitude of the rising and falling slope amplitude. Marziye Eshghi (Speech, Lang. and Hearing Sci., Univ. of North Carolina at Chapel Hill, 002 Brauer Hall, Craniofacial Ctr., Chapel Hill, NC 27599, marziye_eshghi@med.unc.edu), Mohammad Mehdi Alemi (Dept. of Mech. Eng., Virginia Tech, Blacksburg, VA), and Mohammad Eshghi (Inst. of TeleCommun. Systems, Technische Univ. Berlin, Berlin, Germany)

Electrical analog study of the vocal tract has long time ago shown that nasalization can cause a drop of 5–10 dB in the overall vowel amplitude (House and Stevens, 1956). In this preliminary study, the magnitude of rising and falling slope amplitudes from the vowel signal was introduced as a new index to discriminate oral vowels from nasalized vowels. Two speech samples of /iti/ and /iki/ were produced by two normal children and two age-matched children with cleft lip and palate and hypernasal speech. Speech samples were digitally recorded using a microphone and CSL software. The PRAAT software was used to extract the text file of the vowel segment (i.e., from the initiation of the vowel periodicity to the initiation of the constriction interval of the following plosive). Then, the text files were analyzed by MATLAB to measure amplitude rising and falling slopes of the vowel signal. Results indicated that mean falling and rising slopes of the amplitude in the nasalized vowel are smaller than those of the oral vowel. This technique might be considered as an objective index to measure vowel quality in disordered speech studies as well as speech synthesis.

2pSC20. Smiled speech in a context-invariant model of coarticulation. Samuel Akinbo, Thomas J. Heins, Megan Keough, Elise K. McClay, Avery Ozburn, Michael D. Schwan, Murray Schellenberg, Jonathan de Vries, and Bryan Gick (Univ. of Br. Columbia, 2613 West Mall, Vancouver, British Columbia V6T 1Z4, Canada, mkeough@alumni.ubc.ca)

Smiling during speech requires concurrent and often conflicting demands on the articulators. Thus, speaking while smiling may be modeled as a type of coarticulation. This study explores whether a context-invariant or a context-sensitive model of coarticulation better accounts for the variation seen in smiled versus neutral speech. While context-sensitive models assume some mechanism for planning of coarticulatory interactions [see Munhall *et al.*, 2000, Lab Phon. V, 9–28], the simplest context-invariant models treat coarticulation as superposition [e.g., Joos, 1948, Language **24**, 5–136]. In such a model, the intrinsic biomechanics of the body have been argued to account for many of the complex kinematic interactions associated with coarticulation [Gick *et al.*, 2013, POMA **19**, 060207]. Largely following the methods described in Fagel [2010, Dev. Multimod. Interf. **5967**, 294–303], we examine articulatory variation in smiled versus neutral speech to test whether the local interactions of smiling and speech can be resolved in a context-invariant superposition model. Production results will be modeled using the ArtiSynth simulation platform (www.artisynth.org). Implications for theories of coarticulation will be discussed. [Research funded by NSERC.]

2pSC21. Languages across the world are efficiently coded by the auditory system. Christian Stilp and Ashley Assgari (Dept. of Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Independent Component Analysis (ICA) is a powerful method for uncovering statistical structure in natural stimuli. Lewicki (2002 *Nature Neuroscience*) used ICA to examine statistical properties of human speech. Filters that optimally encoded speech were an excellent match for frequency tuning in the cat auditory nerve, leading to suggestions that speech makes

efficient use of coding properties in the mammalian auditory system. However, Lewicki only examined American English, which is neither normative nor representative of the world's languages. Here, fourteen languages were examined (Dutch, Flemish, Greek, Javanese, Jul'hoan, Norwegian, Swedish, Tagalog, Tahitian, Urhobo, Vietnamese, Wari', Xhosa, and Yeyi). Each recording contained speech tokens from a native speaker without any background noise for at least one minute. Maximum likelihood ICA was used to create statistically optimal filters for encoding sounds from each language. These filters were then compared to the same physiological measures analyzed in Lewicki (2002). Languages produced a range of ICA solutions, as expected, but were highly consistent with both statistically optimal filters for American English and physiological measures. Results significantly extend Lewicki (2002) by revealing agreement between response properties of the auditory system and speech sounds from a wide range of languages.

2pSC22. High fidelity analysis of vowel acoustic space. Michael H. Coen (Biostatistics and Medical Informatics, Univ. of Wisconsin-Madison, Madison, WI), Houri K. Vorperian, and Raymond D. Kent (Waisman Ctr., Univ. of Wisconsin-Madison, Waisman Ctr., 1500 Highland Ave., # 427, Madison, WI 53705, vorperian@waisman.wisc.edu)

Vowel acoustic space is often characterized by polygons, whose vertices are determined by summary statistics such as mean values of the formant frequencies of distinct phonemes. The F1-F2 quadrilateral is the most familiar of these. However, using summary statistics to represent formant-frequency data presents fundamental limitations. These data are inherently lossy—summarizing large amounts of data with single values; *mean* itself is a non-robust statistic, highly sensitive to outliers; and even robust statistics ignore *distributional information* within the data, which can vary markedly among different phonemes and age groups. We introduce a new approach characterizing and measuring change in formant spaces statically and developmentally. This approach treats acoustic spaces as *point clouds* of data, in which no information is abstracted or lost. Within this framework, we measure the *spatial overlap* of sets of formant data using an approach combining optimization theory and computational statistics. This provides highly sensitive measures of both similarity and extent of temporal change. This novel approach is robust with respect to outliers, noise, and missing values. It also has a strong intuitive and rigorous mathematical foundation and is easily visualized. Finally, it enables detailed examination of individual phonemes and *clustering* of speakers identifying shared developmental patterns. [Work was supported by NIH grants # R01-DC 006282 & P30-HD03352.]

2pSC23. Shannon entropy predicts the sonority status of natural classes in English. Fernando Llanos (School of Lang. and Cultures, Purdue Univ., 640 Oval Dr., West Lafayette, IN 47907, fllanos@purdue.edu), Joshua M. Alexander (Speech, Lang. and Hearing Sci., Purdue Univ., West Lafayette, IN), and Christian E. Stilp (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

Speech sounds tend to co-occur in the speech stream according to specific combinatory patterns predicted from their sonority status [Parker, S. G. (2002). *Quantifying the sonority hierarchy*. Unpublished doctoral dissertation, University of Massachusetts, Amherst, MA]. This study introduces a measure of spectral complexity, inspired by Shannon entropy, that ranks American English phonemes into a minimal version of the sonority hierarchy: vowels > approximants > nasals > fricatives > affricates > stops. Spectral complexity for every consonant and vowel in the TIMIT database was calculated by first parsing the phonemes into 20-ms segments and computing an FFT. For each short-term FFT, Shannon entropy was computed using the distribution of relative amplitudes (dB) across frequency. Average entropy across the FFTs was used to index spectral complexity for the phonemes, which were then sorted by sonority status. Results of a between-group comparison with spectral complexity as the independent variable and natural class as the dependent variable revealed the existence of six significantly different groups with spectral complexity ranking according to the sonority hierarchy. These findings suggest that Shannon entropy is a reliable acoustic correlate of sonority and may account for the combinatory patterns of co-occurrence of speech sounds in the speech stream.

2pSC24. Acoustic modeling of the perception of place information in incomplete stops. Megan Willi and Brad Story (Univ. of Arizona, P.O. Box 210071, Tucson, AZ 85721, mkittles@email.arizona.edu)

Previous research on stop consonant production found that less than 60% of the stops sampled from a connected speech corpus contained a clearly defined hold duration followed by a plosive release [Crystal & House, JASA 1988]. How listeners perceive the remaining portion of incomplete stop consonants is not well understood. The purpose of the current study was to investigate whether relative formant deflection patterns, a potential model of acoustic invariance proposed by Story and Bunton (2010), is capable of predicting listeners' perceptions of acoustically continuous, voiced stop consonants lacking a canonical hold duration. Listeners were randomly presented a total of 60 voiced stop-consonant VCV stimuli, each 100 ms in duration, synthesized using a computational model of speech production. Stimuli were created using a continuum of 20 equal step constrictions along the length of the vocal tract in three vowel-to-vowel contexts [see Story & Bunton, JSLHR 2010]. Participants listened to the stimuli and performed a forced choice test (i.e., /b-d-g/). The phonetic boundaries predicted by the relative formant deflection patterns and phonetic boundaries obtained by the forced choice test were compared to determine the ability of the acoustic model to predict participants' perceptions. The acoustic and perceptual results are reported. [Work supported by NIH R01-DC011275.]

2pSC25. A spectral filtering method for tracking formants in children's speech. Brad H. Story and Kate Bunton (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

Children's speech is typically characterized by high fundamental frequencies (e.g., 300–600 Hz) that create widely spaced harmonic components, producing an apparent undersampling of the vocal tract transfer function. The purpose of this study is to describe a formant measurement technique based on cepstral analysis that does not require modification of the cepstrum itself or transformation back to the spectral domain. Instead, the spectrum is low-pass filtered with a cutoff point (i.e., cutoff "quefrency" in the terminology of cepstral analysis) to preserve only the spectral envelope. To test the method, speech representative of a 2 to 3 year-old child was simulated with an airway modulation model of speech production. The model includes physiologically-scaled vocal folds, vocal tract, and trachea and generates sound output analogous to a microphone signal. The true formant frequencies can be calculated independently of the output signal and thus provide test cases that allow for assessing the accuracy of the formant tracking algorithm. Formant analysis will also be applied to children's natural speech samples to demonstrate the method. [Work supported by NIH R01-DC011275 and NSF BCS-1145011.]

2pSC26. Measures of tone difficulty. Chilin Shih (East Asian Lang. and Cultures, Univ. of Illinois at Urbana-Champaign, 707 S. Mathews Ave., 2090 Foreign Lang. Bldg., Urbana, IL 61801, cls@illinois.edu)

Measures of difficulty are needed in many real life as well as computer-simulated applications. Such measures for text, math and science have long received academic and industrial attention due to the demands for k-12 instruction and assessment. In recent years, the demands for comparable studies of speech are on the rise given the popularity of on-line second language teaching software and games. The goal of this project is to explore whether the acoustic attributes of Mandarin lexical tones obtained from individual sound files can explain their level of difficulty experienced by second language learners. The study uses monosyllabic tones, thus isolating the task from measurements of complexity in order to focus on acoustics. We recorded sound files that are rich in natural variation using different levels of talker-to-listener distance, used quadratic decomposition to obtain three coefficients that represent each tonal contour, and analyzed their relationship with learners' performance. We will report the difference between native and second-language tone perception. The results have potential applications in speech synthesis: to generate tone tokens with different levels of difficulty for language learners, and different levels of talker-to-listener distance.

2pSC27. Sources of variability in consonant perception and their auditory correlates. Johannes Zaar and Torsten Dau (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, Bldg. 352, Kgs. Lyngby 2800, Denmark, jzaar@elektro.dtu.dk)

Responses obtained in consonant perception experiments typically show a large variability across stimuli of the same phonetic identity. The present study investigated the influence of different potential sources of this response variability. It was distinguished between source-induced variability, referring to perceptual differences caused by acoustical differences in the speech tokens and/or the masking noise tokens, and receiver-related variability, referring to perceptual differences caused by within- and across-listener uncertainty. Two experiments were conducted with normal-hearing listeners using consonant-vowel combinations (CVs) in white noise. The responses were analyzed with respect to the different sources of variability based on a measure of perceptual distance. The speech-induced variability across and within talkers and the across-listener variability were substantial and of similar magnitude. The noise-induced variability was smaller than the above-mentioned contributions but significantly larger than the amount of within-listener variability, which represented the smallest effect. To determine how well the source-induced variability is reflected in different auditory-inspired internal representations (IRs), the corresponding perceptual distances were compared to the distances between the IRs of the stimuli. Several variants of an auditory-spectrogram based IR and a modulation-spectrogram based IR were considered and the importance of the different domains for consonant perception was evaluated.

2pSC28. Double entendre: Embedding a secondary message in pointillistic speech. Gerald Kidd and Christine R. Mason (Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, gkidd@bu.edu)

A dual-purpose application for pointillistic speech (Kidd *et al.*, JASA **126**, EL186–201) will be described. The speech was represented by a time-frequency matrix of pure-tone "points" derived from the original stimulus. The pattern of frequencies and intensities of the points coded the primary message while the phases of the points coded an independent secondary message. For example, the audible and intelligible word "shoes" may also convey the (inaudible, unintelligible) ASCII characters of the phrase "The road goes ever on and on…" by the binary phase-coded bit pattern of the pointillistic representation of "shoes." The success in recovering the secondary message via signal processing was examined while the intelligibility of the primary message in speech recognition experiments was measured. Tradeoffs for accuracy in primary and secondary message transfer/reception were explored for both pointillistic speech and hybrid speech comprising natural and pointillistic representations. Some initial observations about the possible uses and limitations of this approach will be considered. [Supported by AFOSR award FA9550-12-1-0171.]

TUESDAY AFTERNOON, 19 MAY 2015 BALLROOM 4, 1:00 P.M. TO 5:20 P.M.

Session 2pUW

Underwater Acoustics: Historical Perspectives on the Origins of Underwater Acoustics II

David L. Bradley, Cochair
Penn State University, PO Box 30, State College, PA 16870

Thomas G. Muir, Cochair
Applied Research Laboratories, University of Texas at Austin, P/O. Box 8029, Austin, TX 78713

Invited Papers

1:00

2pUW1. The Sound Lab at Fort Trumbull, New London, Connecticut, 1945–1996. Cathy Ann Clark (Sensors & Sonar Systems, NUWCDIVNPT, 1176 Howell St., B1320, R457, Newport, RI 02841, cathy.clark@navy.mil)

The Navy Underwater Sound Laboratory was formed in 1945 when the U.S. Navy merged on-going efforts of Columbia and Harvard Universities to combat the German U-boat threat in the North Atlantic. Prior to that time, the Division of War Research of Columbia, housed in a single building at Fort Trumbull, was sponsored by the National Defense Research Committee (NDRC) while the Harvard Underwater Sound Laboratory was doing similar work in Cambridge, Massachusetts. For the next 50 years, until it was formally closed in 1996, the "Sound Lab" continued to support virtually all aspects of Naval Warfare technology. This talk will attempt to describe the rich scientific culture and technological contributions of the New London Sound Lab.

1:20

2pUW2. Naval oceanography contributions to underwater acoustics—The Cold War era. Robert S. Winokur (None, 15201 Redgate Dr., Silver Spring, MD 20905, robwinok@aol.com)

Underwater acoustics is an integral part of the Navy and is the major enabler for its capabilities in anti-submarine warfare. The importance of oceanography to naval warfare in general and anti-submarine warfare in particular was clearly demonstrated during World War II. Anti-submarine warfare became high technology in the post World War II era, especially during the Cold War. The period from

1960 to the mid 1980s was an important time in the history and evolution of underwater acoustics, building on the experience and important discoveries and research made shortly before, during and immediately after World War II. Naval oceanography programs, large-scale measurement programs and Navy support for research and development were key elements in the rapid expansion of knowledge in the full spectrum of underwater acoustics and understanding the effect of the ocean environment on the development and operation of anti-submarine warfare systems. This paper will provide an overview of the relationship of Naval oceanography to the rapid expansion of knowledge of underwater acoustics during an important period of the Cold War era.

1:40

2pUW3. Contributions to underwater acoustics by the Naval Ordnance Laboratory (NOL). Ira Blatstein (Johns Hopkins Univ., 2190 Chesapeake Harbour Dr. East, Annapolis, MD 21403, blatstein@jhu.edu) and John Tino (none, Adelphi, MD)

This presentation documents the contributions to underwater acoustics by NOL, built in White Oak, MD, in 1946 and operated until closing (as the White Oak Laboratory of NSWC) in 1997. Initial work on mine systems led to R&D on a variety of more sophisticated passive/active acoustic weapon systems for mine warfare, submarines, and ASW aircraft. The increasingly complex mission requirements led to research in a wide range of underwater acoustic disciplines and included development of some key measurement systems, platforms, and facilities across a broad range of frequencies. The measurements obtained by NOL personnel advanced the state of knowledge in underwater acoustic short and long range propagation, surface and bottom acoustic reverberation, sound speed variability with depth, as well as a variety of other research topics. This research led to the development of sensors, materials, target detection systems, and other systems that were applied to a wide variety of acoustic systems. The evolution of this work, and its wide variety of applications, will be discussed.

2:00

2pUW4. Underwater acoustics at the Naval Air Development Center. Thomas B. Gabrielson (Penn State Univ., PO Box 30, State College, PA 16804, tbg3@psu.edu)

Much of the progress in underwater-acoustic research from World War II to the present has been funded by government agencies for development of sonar systems. Although sonar is often considered the domain of surface ships and submarines, the naval aviation community has made significant contributions to basic underwater-acoustic measurement and to acoustic-system development. In the 1940s, the U.S. Navy established the Naval Air Development Center (NADC) in Warminster, Pennsylvania, as the lead laboratory for naval aviation research and development and a substantial part of the work of that laboratory supported development of air-deployed sonar systems. Partnerships between NADC, other laboratories both domestic and foreign, and manufacturers produced a stream of innovative, inexpensive, and expendable devices to support the missions of marine patrol aircraft; these same devices were used extensively for ocean-acoustic measurements. Basic bottom-reflection loss, ambient-noise level and directivity, and reverberation measurements were made using air-deployed sonobuoys and acoustic sources. While lacking the precision of ship-based measurements, the cost of airborne surveys was low, deployment was rapid, and the coverage was ultimately global.

2:20

2pUW5. Pioneers in side scan sonar: Julius Hageman and the shadowgraph. Kerry Commander and Daniel Sternlicht (Sci. and Technol. Dept., Naval Surface Warfare Ctr. Panama City Div., NSWC PCD, Code X, 110 Vernon Ave., Panama City, FL 32407-7001, kerry.commander@navy.mil)

The concept of the side scan sonar was developed during the early 1950s at the U.S. Navy Mine Defense Laboratory, Panama City Florida—now known as the Naval Surface Warfare Center Panama City Division. In technical reports and laboratory notebooks, Dr. Julius Hageman, a German scientist who relocated to the Laboratory after World War II and worked there until his death in 1964, outlined the proposed "short-range high-definition mine location sonar" that would eventually become the C-MK-1 mine classification sonar system, more commonly known as "Shadowgraph." Hageman's patent for the concept (US Patent 4,197,591) was first disclosed in 1958, but remained classified until finally issued in 1980. The Shadowgraph was contracted by the U.S. Navy in 1957 and towed primarily from Oceangoing Mine Sweepers. It was operated as an undersea search and survey tool for more than 25 years before decommissioning in 1991. The Shadowgraph was a 1.5 MHz dual-sided side scan sonar, with range of 100 feet, and imaging resolution of approximately 3 in. square at a range of 75 feet. During its service life, it located numerous lost objects and aircraft on the sea floor and to this day has influenced the development of commercial and military sonars.

2:40

2pUW6. A few Canadian contributions to underwater acoustics. Harold M. Merklinger and John C. Osler (DRDC - Atlantic Res. Ctr., PO Box 1012, Dartmouth, Nova Scotia B2Y 3Z7, Canada) (, john.osler@drdc-rddc.gc.ca)

An historical perspective on four Canadian achievements in underwater acoustics will be presented: Towed Variable Depth Sonar (TVDS); Chapman-Harris model for reverberation; seabed-interaction in shallow water, and uninhabited underwater vehicles (UUVs) for Arctic acoustics. The poor performance of hull-mounted sonars in Canadian waters during WWII prompted TVDS development in 1947. The resulting prototype (CAST/1X) was tested against UK and US developmental systems in 1958. Subsequently, the UK purchased the Canadian system and the US "re-packaged" their sonars according to the Canadian design. An understanding of the effects of back-scattering from the sea surface and bottom was required to model sonar system performance. Experiments began circa 1960 leading to a new model for sea surface back-scattering, and revealing the ocean volume as an additional factor. The conditions governing propagation and noise in shallow waters remained mysterious in the 1970s, until the significant role of the seabed emerged through theoretical developments (e.g., loss due to shear-waves) and experiments conducted over a variety of seabed types. Political, sovereignty and energy issues prompted Canada to investigate under-ice acoustics in the Arctic beginning circa 1957. The efforts ultimately resulted in the ability to undertake under-ice acoustic and geophysical surveys using long endurance UUVs.

3:00–3:20 Break

3:20

2pUW7. Early acoustics research at the Navy's Pacific research and development laboratory. C. D. Rees (S&T Forecasting, Assessment, and Transition (72120), SSC Pacific, 53560 Hull St., Code 72120, San Diego, CA 92152-5001, Dave.Rees@navy.mil)

Few organizations have a more essential connection to underwater acoustics than the US Navy; the Navy's mission success has been critically dependent on understanding and exploiting underwater acoustics since the advent of submarines. In its several manifestations since WW II, the Navy's San Diego-based R&D laboratory (currently SPAWAR Systems Center Pacific) has played an important role in the development of the underwater acoustics field. This progress has been in conjunction with other San Diego institutions, such as Scripps Institute of Oceanography and the Marine Physics Laboratory, and other Navy laboratories. We provide a historical overview of acoustics research at SSC Pacific and predecessors in the era following WW II, including the impact of the environment, Arctic acoustics, fielded systems, sonar and active acoustics, and marine mammal acoustics, and the scientists and researchers such as Homer Bucker, Sam Ridgeway, Shelby Sullivan and others who worked in developing our modern understanding of underwater acoustics on the edge of the Pacific.

3:40

2pUW8. History of underwater electroacoustic transducer standards, calibration methods, facilities, and some early contributors. David A. Brown (ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net) and Anthony Paolero (Code 153, Naval Underwater Warfare Ctr., Newport, RI)

The practice of realizing underwater acoustic standards and methods for the systematic calibration of the wide variety of electroacoustic transducers is rich in tradition and has given rise to some unique facilities. In about 1940, the US Navy Office of Scientific Research and Development established a center for Underwater Acoustic Metrology under the expertise of Columbia University in collaboration with Bell Laboratories, which subsequently resulted in the establishment of the Underwater Sound Research Laboratory (USRL) in Orlando Florida with unique acoustic lake testing facilities. That facility later became a detachment of the Naval Research Laboratory (USRD) and was the home for a concerted effort in transducer development and calibration for many years. The National Bureau of Standards and later the National Institute for Standards (NIST) deferred the activity of establishing and maintaining underwater acoustic standards and calibration methods to the USRD Navy facility. This paper summarizes some of the transducers standards, methods and facilities and many of the important contributions of early pioneers in this field including Ira Groves, Robert Bobber, Joseph Blue, and others. USRD continues today as a division at NUWC, Newport.

4:00

2pUW9. Early history of underwater acoustics research at SACLANTCEN, La Spezia, Italy. Finn B. Jensen (NATO CMRE, Viale San Bartolomeo 400, La Spezia 19126, Italy, finn.jensen@cmre.nato.int)

The SACLANT ASW Research Centre was established in 1959 to provide scientific and technical advice to the Supreme Allied Commander Atlantic (SACLANT) in the field of antisubmarine warfare and to respond to the needs of NATO nations and maritime commands. Hence, it was a NATO-sponsored research center placed in the Italian Navy compound in La Spezia, Italy, and it was internationally staffed (total about 230) with 50 rotational scientists coming from NATO nations on both sides of the Atlantic. This review covers three decades of achievements in ocean acoustics research from the foundation in 1959 to the end of the Cold War in 1989. Both basic and area-related experimental studies of propagation in deep and shallow waters, ambient noise, signal coherence, and reverberation were conducted over the years. From the early 1970s, the Centre also initiated the development of a set of high-fidelity acoustic models, which subsequently were made available to researchers within the NATO community. Finally, the Centre served as a focus for international cooperation by organizing scientific conferences and workshops on a yearly basis and by conducting multinational measurement programs in the Mediterranean, the Black Sea, and the Eastern North Atlantic from Gibraltar up to the Barents Sea.

4:20–5:20 Panel Discussion

Meeting of Accredited Standards Committee (ASC) S3/SC 1, Animal Bioacoustics

D. K. Delaney, Chair ASC S3/SC 1
USA CERL, 2902 Newmark Drive, Champaign, IL 61822

D. S. Houser, Vice Chair ASC S12
National Marine Mammal Foundation, 2240 Shelter Island Drive Suite 200, San Diego, CA 92106

Accredited Standards Committee S3/SC 1 on Animal Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43/SC 1 Noise and ISO/TC 43/SC 3, Underwater acoustics, take note-those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 19 May 2015.

Scope of S3/SC 1: Standards, specifications, methods of measurement and test, instrumentation, and terminology in the field of psychological and physiological acoustics, including aspects of general acoustics, which pertain to biological safety, tolerance, and comfort of non-human animals, including both risk to individual animals and to the long-term viability of populations. Animals to be covered may potentially include commercially grown food animals; animals harvested for food in the wild; pets; laboratory animals; exotic species in zoos, oceanaria or aquariums; or free-ranging wild animals.

TUESDAY AFTERNOON, 19 MAY 2015 DUQUESNE, 3:00 P.M. TO 4:15 P.M.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

C. J. Struck, Chair ASC S3
CJS Labs, 57 States Street, San Francisco, CA 94114 1401

P. B. Nelson, Vice Chair ASC S3
Department of SLHS, University of Minnesota, 115 Shevlin, 164 Pillsbury Drive S.E., Minneapolis, MN 55455

Accredited Standards Committee S3 on Bioacoustics. Working group chairs will report on the status of standards under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAGs for ISO/TC 43 Acoustics and IEC/TC 29 Electroacoustics, take note-those meetings will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 19 May 2015.

Scope of S3: Standards, specifications, methods of measurement and test, and terminology in the fields of psychological and physiological acoustics, including aspects of general acoustics which pertain to biological safety, tolerance, and comfort.

Meeting of Accredited Standards Committee (ASC) S12 Noise

W. J. Murphy, Chair ASC S12
NIOSH, 4676 Columbia Parkway, Mail Stop C27, Cincinnati, OH 45226

S. J. Lind, Vice Chair ASC S12
The Trane Co., 3600 Pammel Creek Road, Bldg. 12-1, La Crosse, WI 54601 7599

Accredited Standards Committee S12 on Noise. Working group chairs will report on the status of noise standards currently under development. Consideration will be given to new standards that might be needed over the next few years. Open discussion of committee reports is encouraged.

People interested in attending the meeting of the TAG for ISO/TC 43/SC 1 Noise, take note-this meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 19 May 2015.

Scope of S12: Standards, specifications, and terminology in the field of acoustical noise pertaining to methods of measurement, evaluation, and control, including biological safety, tolerance and comfort, and physical acoustics as related to environmental and occupational noise.

TUESDAY EVENING, 19 MAY 2015 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday and Thursday evenings. On Tuesday, the meetings will begin at 7:30 p.m., except for Engineering Acoustics, which will hold its meeting starting at 4:30 p.m. On Thursday evening, the meetings will begin at 8:00 p.m. or 8:30 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 (4:30 p.m.)	Brigade
Acoustical Oceanography	Rivers
Animal Bioacoustics	Brigade
Architectural Acoustics	Ballroom 3
Physical Acoustics	Ballroom 4
Psychological and Physiological Acoustics	Commonwealth 2
Structural Acoustics and Vibration	Commonwealth 1

Session 3aAAa

Architectural Acoustics, ASA Committee on Standards, and Noise: Public Policy Implementation of School Acoustics

Jason E. Summers, Chair
Applied Research in Acoustics LLC, 1222 4th Street SW, Washington, DC 20024-2302

This moderated panel discussion brings together members of the Society involved in the development and adoption of the classroom-acoustics standard (ANSI S12.60-2002) and its adoption by local and state governments with representatives from the education-policy community and those familiar with the facilities and infrastructure challenges of local school districts. Following a brief introduction from the chair, Society member David Lubman will speak briefly on his experiences working with the Society and federal agencies in the technical development of the standard. Society member Bennett M. Brooks will speak to his experience working with other interested citizens as an advocate for the adoption of the standard in Connecticut. They will be joined by representatives from local schools and foundations that support education who will address facilities and infrastructure challenges and the relationship of this issue to the broader concerns of education policy. After remarks and moderated discussion by the panelists, the session will be open to questions from the audience.

Session 3aAAb

Architectural Acoustics and Noise: Session in Honor of Dick Campbell

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

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

Chair's Introduction—10:25

Invited Papers

10:30

3aAAb1. Remembering the life and contributions of Dick Campbell, a man of many talents. Eric L. Reuter (Reuter Assoc., LLC, 10 Vaughan Mall, Ste. 201A, Portsmouth, NH 03801, ereuter@reuterassociates.com)

Aside from his many technical contributions to audio and acoustics, Dick Campbell was a teacher and mentor to many, and had a diverse set of interests outside of his scientific pursuits. A horseman, pilot, sailor, marina owner, musician, tinkerer, community leader, and devoted father, he never had any trouble staying busy, even when faced with some harsh challenges. The author was a student of Dick's at Worcester Polytechnic Institute, and greatly valued his enthusiastic mentoring. This biographical presentation will include several personal anecdotes, as well as some stories and surprises collected from Dick's family and his many friends and colleagues.

10:50

3aAAb2. A series of special sessions on architectural acoustics and audio. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com) and Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, Lowell, MA)

Along with his love of concert halls, opera houses, and pipe organs that are often considered the top of the architectural acoustics food chain, Dick Campbell always shared his enthusiasm for smaller rooms, loudspeakers, computers, signal processing, and all forms of media. Daring to incorporate MIDI-driven, synthesized sounds produced through loudspeakers into the pit orchestra, Dick knew how to rabble rouse—and inspire. One legacy of his politely provocative thinking is a series of 11 special sessions—so far—started with him at the 1999 Columbus meeting, and continued to this day by the authors. The thread of related special sessions had its most recent iteration at the Fall 2014 meeting in Indianapolis, "Architectural acoustics and audio—Rooms, systems, and techniques for adapting, enhancing, and fictionalizing acoustic traits", with 22 papers covering a wide range of ideas and concepts. A tour of these sessions' past topics and authors reveals a rich body of work that might otherwise have been overlooked in ASA meetings without the seed planted by Dick Campbell.

11:10

3aAAb3. My time with Dr. "Dick" Campbell. William R. Michalson (Dept. of Elec. and Comput. Eng., Worcester Polytechnic Inst., 100 Inst. Rd., Worcester, MA 01609, wrm@wpi.edu)

I have been asked by a former student if I would have any interest in speaking about my experiences with Dick Campbell as a colleague and friend during our time working together. My answer? Of course, I would be honored to speak about the unofficial "Mayor of Woods Hole." We met shortly after I joined WPI. Dick's interest in acoustics and my interest in music electronics began a working relationship that evolved into a years-long friendship. Our collaborations benefited (and annoyed) the students we co-advised. Dick was bedridden when we last worked together, but that was Dick—have laptop, will Skype. He never seemed to let the "little things" get in the way of anything that grabbed his interest. I will talk about some of the projects we worked on together, including acoustic modeling, identification of mosquito species by their acoustic signature, and others. Additionally, I'll talk about the times I spent in his office at WPI, the "acoustics lab" in the Higgins garage, and his laboratory in Woods Hole—the World headquarters for "Bang-Campbell." Thank you for your attention, and thank you Dick. You've been an extraordinarily influential person in my life and the lives of many others.

11:30–12:00 Open Forum Contributions Invited and Welcomed

WEDNESDAY MORNING, 20 MAY 2015 — KINGS 2, 8:00 A.M. TO 11:45 A.M.

Session 3aBA

Biomedical Acoustics: Acoustic Radiation Force in Biomedical Applications I

Mostafa Fatemi, Cochair
Physiology & Biomedical Engineering, Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905

Matthew W. Urban, Cochair
Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905

Chair's Introduction—8:00

Invited Papers

8:05

3aBA1. Acoustic radiation force in biomedical applications: origin, past, present, and future. Armen Sarvazyan (Artann Labs., 1459 Lower Ferry Rd., Trenton, NJ 08618, armen@artannlabs.com)

In this talk, an overview of history and physical basis of biomedical applications of acoustic radiation force with a further look into new developments in this field will be presented. In 1902, Lord Rayleigh published his classical work on the theory of sound, introducing the concept of acoustic radiation pressure. Experimental demonstration of radiation force acting on particles in the standing wave field was made by August Kundt (1874). Much later, Wood and Loomis built radiation force balance (1927). Detailed analysis of ultrasound radiation force related to biomedical applications was made in numerous reviews and original articles of Wes Nyborg in the

1960s and 1970s. In spite of over century-old history, most of the significant biomedical applications of acoustic radiation force became known and were extensively studied only during last couple of decades. We will present and discuss recent progress in numerous applications such as the elasticity imaging, assessing viscoelastic properties of biological tissue, monitoring lesions during therapy, the manipulation of cells in suspension, acoustical tweezers, targeted drug and gene delivery. In addition to well established applications, the acoustic radiation force has considerable potential in numerous new areas, which will be discussed in the talk. [NIH R21AR065024.]

8:25

3aBA2. Direction of acoustic radiation force on spherical scatterers in soft tissue. Benjamin C. Treweek, Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, btreweek@utexas.edu)

A theory for acoustic radiation force on a viscoelastic sphere of arbitrary size in soft tissue has been reported previously for a nonaxisymmetric incident field described via spherical harmonic expansion [Ilinskii *et al.*, POMA **19**, 045004 (2013)]. At the Fall 2014 ASA meeting, the model was used to compute the radiation force on scatterers with different sizes and properties at various positions relative to the focus of an axisymmetric incident beam. For a particle located away from the focus, the model predicts a change in the direction of the axial or the transverse component of the radiation force depending on properties of both the particle and the host medium. The focus of the present contribution is this change in direction. Scatterers with various sizes and mechanical properties are considered, and small particles are found to be more prone to this phenomenon. Additionally, the reversal in direction is found to be sensitive to variations in the shear modulus of the host medium. Comparisons are made with liquid as the shear modulus of the host medium spans the range of values encountered in soft tissue. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

8:45

3aBA3. Radiation torque, streaming, and viscous absorption due to orthogonal waves in slightly viscous fluids. Likun Zhang (Ctr. for Nonlinear Dynam., Univ. of Texas at Austin, 2515 Speedway, Stop C1610, Austin, TX 78712-1199, lzhang@chaos.utexas.edu) and Philip L. Marston (Dept. of Phys. and Astronomy, Washington State Univ., Pullman, WA)

Torque generated by orthogonal waves in slightly viscous fluids is analyzed to explore its connection with the near-boundary viscous absorption and streaming [Zhang and Marston, J. Acoust. Soc. Am. **131**, 2917–2921 (2014)]. The analysis is based on a viscous correction and beam superposition of vortex fields. The analysis reports an approximation for the torque on a compressible small solid sphere and the viscous power absorption [Eqs. (15) and (21) in Zhang and Marston]. The torque expression in the dense sphere limit recovers prior expressions using near-boundary flow and streaming analyses: [Busse and Wang, J. Acoust. Soc. Am. **69**, 1634–1638 (1981)] and [Lee and Wang, J. Acoust. Soc. Am. **85**, 1081–1088 (1989)]. Our analysis extends prior results to give the torque on small compressible solid spheres of finite mass. The torque does not depend explicitly on the compressibility of the sphere, but does depend on the density of the sphere. Our analysis also gives the proportional relationship between the torque and the near-boundary viscous absorption. The results are generalized to cases of standing orthogonal waves and acoustic vortex fields, and other small axisymmetric obstacles such as circular cylinders and disks. [Work supported by the 2013-14 F. V. Hunt Postdoctoral Fellowship (Zhang) and by ONR (Marston).]

9:05

3aBA4. The pull of finite acoustic beams. Farid G. Mitri (Area 52 Technol., Chevron, 5 Bisbee Court, Santa Fe, NM 87508, f.g.mitri@ieee.org)

The counterintuitive effect of pulling an object back towards a finite (bounded) circular source with the use of acoustic forces of progressive waves is demonstrated, which defines the acoustic "tractor beam" behavior [F. G. Mitri, "Near-field single tractor-beam acoustical tweezers," Appl. Phys. Lett. **103**(11), 114102, 2013; "Single Bessel tractor-beam tweezers," Wave Motion **51**(6), 986–993, 2014]. Experimental results and numerical predictions show the emergence of negative forces due to continuous (CW) and amplitude-modulated (AM) propagating waves. The circumstances under which a collimated beam emanating from a finite aperture can act as a tractor beam and related investigations are also discussed.

9:25

3aBA5. Applications of acoustic radiation force for microvascular tissue engineering. Diane Dalecki, Eric S. Comeau (Dept. of Biomedical Eng., Univ. of Rochester, 310 Goergen Hall, P.O. Box 270168, Rochester, NY 14627, ddalecki@ur.rochester.edu), and Denise C. Hocking (Pharmacology and Physiol., Univ. of Rochester, Rochester, NY)

We have developed a non-invasive ultrasound-based method to spatially pattern cells within three-dimensional (3D) hydrogels, and have demonstrated translation of this technology to microvascular tissue engineering. Acoustic radiation forces associated with an ultrasound standing wave field (USWF) can rapidly organize a variety of cell types into distinct spatial patterns within 3D hydrogels. USWF-induced patterning of endothelial cells into distinct multicellular planar bands can lead to the rapid formation of complex microvessel networks throughout the volume of the collagen hydrogel. Recent efforts have focused on optimizing acoustic exposure parameters to control resultant 3D microvessel morphology, producing multilayered composite constructs with complex, physiologically relevant vascular morphologies, and developing dual ultrasound transducer systems to noninvasively direct cell patterning within injectable collagen hydrogels in situ. Engineering microvessel networks that structurally and functionally mimic native microvasculature is critical for the fabrication and survival of a broad range of bioengineered tissues, and for the fabrication of small, vascularized tissue constructs for use as realistic cost-effective *in vitro* models for drug discovery and testing.

9:45–10:00 Break

Contributed Papers

10:00

3aBA6. Acoustic radiation force and particle dynamics in cylindrical resonators. Lev A. Ostrovsky (PSD, NOAA ESRL, 325 Broadway, R/PSD99, Boulder, CO 80305, lev.a.ostrovsky@noaa.gov)

Acoustic radiation force is widely used in biomedical studies and medical diagnostics. Among the promising areas of its application is concentration and stirring of particles and bubbles in ultrasonic resonators (e.g., [1]). In a number of cases, cylindrical resonators have a practical advantage vs. plane resonators (acoustic interferometers) due to the axial energy concentration [2], [3]. Theoretical analysis of particle dynamics in such resonators is often very complicated. This presentation outlines recent results in modeling of dynamics of micro-particles and microbubbles in cylindrical resonators. In particular, concentration and separation of "heavy" and "light" particles and near-resonant bubbles are discussed. The role of ultrasound nonlinearity in the considered effects is also discussed. [1]. A. Sarvazyan and L. Ostrovsky, J. Acoust. Soc. Am. **125**, 3548–3554 (2009). [2]. A. Priev and A. Sarvazyan, J. Acoust. Soc. Am. **125**, 2593 (2009). [3]. L. Ostrovsky, A. Priev, V. Ponomarev, and Y. Barenholz, Proc. Meetings Acoust. **14**, 020002 (2013).

10:15

3aBA7. Acoustic radiation force on a particle in ultrasonic standing wave fields driven by a piezoelectric plate. Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA), Ben Ross-Johnsrud, Minghe Liu (FloDesign Sonics, 380 Main St., Wilbraham, MA 01095, b.johnsrud@fdsonics.com), Yurii Ilinskii, and Evgenia Zabolotskaya (Appl. Res. Labs, The Univ. of Texas at Austin, Austin, TX)

Acoustic radiation forces on a particle or droplet in a liquid have been studied mostly for relative simple acoustic fields such as plane traveling waves, or planar standing waves. In this study, a more complicated acoustic field is considered, namely, the acoustic standing wave field generated by a rectangular piezoelectric plate with finite size. The piezo-electric plate is excited in its thickness mode. A three dimensional model of the vibration of the piezo-electric plate has been developed. The plate is loaded by an acoustic standing wave field in a fluid on one side, and is air backed on the other side. The three-dimensional acoustic standing wave field generated by the piezo-electric plate is calculated. The acoustic radiation force on a particle or droplet can then be calculated using a framework such as developed by Gor'kov or Ilinskii and Zabolotskaya. The theoretical calculations are compared with numerical predictions using a two-dimensional model of the piezo-electric plate and resonator. Finally, experimental results were obtained in a resonator with a one inch acoustic path length and driven by a 2 MHz PZT-8 one inch by one inch piezo-electric plate. Experimental results will be compared with theoretical and numerical predictions.

10:30

3aBA8. Enhancement of osteogenesis and stem cell differentiation by injectable nanocomposite and dynamic acoustic radiation force stimulation. Vaishnavi Shrivastava, Sunny Patel, Minyi Hu, Suphannee Pongkitwitoon, Balaji Sitharaman, and Yi-Xian Qin (Biomedical Eng., Stony Brook Univ., BioEng. Bldg., Rm. 215, Stony Brook, NY 11794, yi-xian.qin@stonybrook.edu)

Osteopenia affects mineral density, microstructure, and integrity of bone, leading to increased risk of fractures, as well as high rates of non-union, which affect patients' quality of life. Current treatments are ineffective, requiring invasive surgeries and/or long-term drug therapy. The objective of this study was to develop a novel noninvasive biomimetic treatment for rapid regeneration to promote cell differentiation and osteogenesis. An injectable orthopedic implant was designed by developing a thermosensitive scaffold incorporating carbon nanotubes and chitosan-β glycerophosphate hydrogels. An innovative biophysical stimulation using dynamic ultrasound radiation force (ARF) was used to induce carbon nanotube resonance for regulating osteogenic differentiation of stem cells. An assay on activity of ALP, a biomarker of osteogenesis, and a fluorescence-based live/dead cell assay were conducted to determine the best treatment for inducing rapid cell formation. The single-walled carbon nanotube scaffold and 60 mW/cm^2 ARF group was the best treatment, and enhanced ALP activity and stem cell differentiation by 430%. Cell viability was increased by 65% through the ARF treatment. This study developed a novel treatment for regulating cell proliferation using single-walled carbon nanotube nanocomposite implants and synergistic stimulation of stem cells with ultrasound. This rapid, noninvasive, cost-effective treatment may provide an innovative alternative for osteoporotic treatment and rapid fracture healing.

10:45

3aBA9. Acoustic separation of milk fat globules—Principles in large scale processing. Thomas Leong, Linda Johansson (Faculty of Sci., Eng. and Technol., Swinburne Univ. of Technol., John St. Hawthorn, Melbourne, Victoria 3122, Australia, tleong@swin.edu.au), Pablo Juliano (CSIRO Food and Nutrition, Melbourne, Victoria, Australia), and Richard Manasseh (Faculty of Sci., Eng. and Technol., Swinburne Univ. of Technol., Melbourne, Victorai, Australia)

The acoustic manipulation of particulates in standing wave fields is a well-established technique, with the fundamental theory describing the acoustic radiation forces first reported in 1934 by King. To date, there are few demonstrations of the technique on a volume-scale relevant to industrial application, due to difficulties in scaling the acoustic radiation forces over large distances. Other issues such as the onset of acoustic streaming with high acoustic power input may also impact upon the effectiveness of separation in large systems by disrupting the collection of particulates in the pressure nodal/antinodal regions. In recent work, we have demonstrated the capability of ultrasonic standing waves applied in a liter-scale vessel to concentrate and/or remove fat globules from whole milk with volume throughputs up to 30 L/h. By tuning parameters according to acoustic fundamentals, the technique can be used to specifically select milk fat globules of different sizes in the collected fractions. We report key design and operation principles at large scale using milk as a model system, which has potential to be applied to particulate fluids of biomedical relevance such as blood and lipids.

11:00

3aBA10. Understanding diffractive and reflective contributions to scattering and extinction of intersecting plane waves. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu)

Scattering and extinction of intersecting plane waves is relevant to many processes of interest, including the interpretation of bistatic scattering and the understanding of acoustical radiation forces. A special case that exhibits diverse behavior is the illumination of a sphere by an acoustic Bessel beam. Consider high frequency scattering in which the sphere radius exceeds the wavelength. Recent progress in understanding diffractive scattering contributions uses Babinet's principle and the Kirchhoff approximation [P. L. Marston, J. Acoust. Soc. Am. **135**, 1668–1671 (2014)]. The optical theorem for invariant beams [L. Zhang and P. L. Marston, J. Acoust. Soc. Am. **131**, EL329–EL335 (2012)] was used to evaluate the extinction. Comparison with the partial wave series for the extinction supports the utility of quasi-scaling involving the Bessel beam's conic angle. The reflective scattering contribution is subtler; however, evaluating it quantitatively explains an abrupt transition in the spacing of fringes in bistatic scattering patterns for spheres centered in Bessel beams. The transition is visible where the scattering angle passes through the beam's conic angle [P. L. Marston, J. Acoust. Soc. Am. **121**, 753–758 (2007)]. Diffractive and reflective contributions for smooth objects illuminated by intersecting waves are relevant to many other situations. [Work supported by ONR.]

11:15

3aBA11. Spatially uniform harmonic acoustic radiation force excitation using one-dimensional linear array. Mahdi Bayat, Azra Alizad, and Mostafa Fatemi (Physiol. and Biomedical Eng., Mayo Clinic, 200 1st St. SW, Rochester, MN 55905, bayat.mahdi@mayo.edu)

Low frequency harmonic force excitation can be used for both imaging and tissue characterization. This type of excitation can be achieved via

intersecting two beams that slightly differ in frequency. One requirement of this technique is creating focused intersecting beams using available hardware. Previous studies have suggested different configurations for creating harmonic radiation force using linear arrays. However, these methods rely on identical frequency response assumption across all elements of the aperture. In practice, this assumption can be easily violated, especially if the transmit frequencies are not around the center frequency of the probe. When using electronic steering and focusing at different locations, this can result in a non-uniform radiation force pattern due to frequency changes in the sub-aperture. When used for vibro-acoustography, this manifests as vertical streaks in the final image that can severely degrade the quality of the image. We present a novel technique that minimizes element transmit frequency switching which in turn results in a more uniform excitation intensity. The results of applying this aperture configuration for vibro-acoustography imaging of breast phantoms with hard inclusions show significant improvements over previous methods. [This work was supported by NIH Grant R01 EB17213.]

11:30

3aBA12. Sternal vibrations reflect hemodynamic changes during immersion: Underwater ballistocardiography. Andrew D. Wiens, Andrew Carek, and Omer T. Inan (Elec. and Comput. Eng., Georgia Inst. of Technol., 85 Fifth St. NW, Atlanta, GA 30308, andrew.wiens@gatech.edu)

Ballistocardiography (BCG) is a method for measuring the small vibrations of the body caused by the beating of the human heart. In this study, vibration measurements of the sternum for the purpose of noninvasive hemodynamic monitoring during total body immersion in water are recorded and examined for the first time. Three individuals wore a low-noise accelerometer while immersed in water of varying temperature up to the neck, and Valsalva maneuvers were performed. The resulting waveforms reveal distinct differences in signal morphology between three postures and two water temperatures, suggesting that underwater BCG could be applied in aquatic environments without a need for electrodes.

WEDNESDAY MORNING, 20 MAY 2015 KINGS 1, 8:30 A.M. TO 11:55 A.M.

Session 3aED

Education in Acoustics and Physical Acoustics: Preparing Graduate Students for Careers in Acoustics

Daniel A. Russell, Cochair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

E. Carr Everbach, Cochair
Engineering, Swarthmore College, 500 College Avenue, Swarthmore, PA 19081

Chair's Introduction—8:30

Invited Papers

8:35

3aED1. Careers in acoustics: A journey from research lab engineer to tenure-track faculty. Andrew R. Barnard (Mech. Eng. - Eng. Mech., Michigan Technolog. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, arbarnar@mtu.edu)

As they approach the end of their Bachelor's or Master's degree, many students ask themselves "Should I get my Ph.D. or take the money and move to industry?" There is a perception that it may be more difficult to get a job after obtaining a Ph.D. due to the dreaded "over-qualification." I have found the opposite in my career. In this talk, I will present my story of completing my Ph.D. and moving into a position at a university research laboratory, the Applied Research Lab at Penn State, and my subsequent transition into a tenure-track faculty position in the department of Mechanical Engineering—Engineering Mechanics at Michigan Tech University. Pros and Cons of both positions will be discussed as will my view on the importance of the Ph.D. researcher in the future of academia, government, and industry. I will conclude with my advice for current and future Ph.D. students interested in research and/or academics.

8:55

3aED2. Perspective from a college of engineering at a large, public research university. Anthony A. Atchley (Gradate Program in Acoust., Penn State Univ., College of Eng., 101C Hammond Bldg., University Park, PA 16802, atchley@psu.edu)

A career at a large, public research-intensive university is the gateway to a life-long opportunity to create new knowledge, develop realistic solutions to the world's greatest challenges, influence young minds, educate the next generation of leaders, and service society. The value proposition for such a privileged career opportunity, however, requires a life-long commitment to excellence, self-improvement, and reinvention. This presentation, based upon the author's experience as a faculty member, graduate program chair, and senior college-level administrator, is aimed at students who are considering faculty positions at such institutions. It will address getting the most out of your Ph.D. program and post-doctoral experience; the critical first few years as a faculty member; balancing the demands of and integrating teaching, research, and service; becoming skilled at writing proposals and desseminating research results; qualities you should seek in an institution you are considering and questions you should ask when interviewing.

9:15

3aED3. Publish or perish and funding or failure—The dark side of a career in academia. Ralph T. Muehleisen (Energy Systems, Argonne National Lab., 9700 S. Cass Ave., Bldg. 221, Lemont, IL 60439, rmuehleisen@anl.gov)

A career in academia can be extremely rewarding. It can be intellectually stimulating, financially rewarding, provide an extremely flexible work schedule, and give an opportunity for one to truly impact the world in a positive way. But, there is a dark side to academia that one should be aware of before jumping in. Academia can also be full of petty and insignificant politics, enormous egos, and unrealistic expectations. Perhaps the darkest side relates to the need to "publish or perish" and "find funding or fail" that can be found at many research universities. This presentation will provide an honest discussion of the ups and downs of academia from someone who found both success and failure.

9:35

3aED4. Navigating the track to tenure: A liberal arts college perspective. David T. Bradley (Phys. + Astronomy, Vassar College, 124 Raymond Ave., #745, Poughkeepsie, NY 12604, dabradley@vassar.edu)

Pursuing a career in the academy can be rewarding and challenging. Securing a tenure-track position at a liberal arts college can be particularly difficult since there are far fewer available positions, and because this type of institution is very different from the ones that grant the requisite Ph.D. degree for the job. A liberal arts college is an institution that is undergraduate focused, offering a breadth of courses in the humanities, social sciences, and natural sciences, while still providing the appropriate depth and rigor to prepare students for graduate school and industry. These colleges tend to have a stronger emphasis on teaching than larger universities, although professors are still expected to develop a thriving research program, typically only with the help of a few undergraduate research assistants. This presentation will discuss the unique aspects of being a professor at a liberal arts college while describing one successful path to the coveted tenured status.

9:55–10:10 Break

10:10

3aED5. Preparing for a career in academia: Managing students in research. Kent L. Gee, Tracianne B. Neilsen, Scott D. Sommerfeldt, and Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

A new faculty member faces challenges associated with meeting and balancing various teaching, research, and citizenship demands. This includes managing students as part of developing a research program. Despite the vital importance of this skill, effective employee management is not something a student inherently learns in graduate school nor does it often receive attention as part of new faculty development workshops. This presentation discusses lessons learned regarding research student management. These include setting a scholarly goal at the outset with specific result-driven milestones and clear expectations of the "end game," adopting a management style that is best suited to each student's personality, adapting the project where possible to student strengths, and helping them learn to write as early as possible. Graduate students can be trained to become effective peer mentors of undergraduate students, increasing both a sense of teamwork and overall productivity. New faculty members will benefit from actively seeking mentorship by more experienced colleagues who have successfully built student-based research programs.

10:30

3aED6. An administrative perspective on preparing for careers in academia. Scott D. Sommerfeldt (Dept. of Phys., Brigham Young Univ., N181 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

As the dean of a college, one of my responsibilities is to interview each person that the departments in my college have some interest in hiring. As a result, I have interviewed dozens of candidates, some of whom were very strong and some of whom were far weaker. In this presentation, I will overview some aspects of career preparation that students can focus on that will help prepare them to be strong candidates if they are interested in pursuing a career in academia. I will also discuss some aspects of the interviewing process that students can prepare for in an effort to present themselves favorably during interviews.

10:50

3aED7. Preparing University of Mississippi graduate students for careers in acoustics. Richard Raspet (National Ctr. for Physical Acoust, NCPA, Univ. of MS, University, MS 38677, raspet@olemiss.edu)

The acoustics program at the University of Mississippi originated in the Physics Department and is still closely associated with physics. The physics department requires all students teach two semesters of laboratory under the supervision of a laboratory physicist. The required physics curriculum is extensive so that the acoustics students only receive two semesters of acoustics courses. This leads to careful course design so that our students are broadly literate in acoustics. The necessary compression of class work means that speaking and writing skills are taught to the student by their research advisor in preparing conference talks, written papers, and dissertations.

11:10–11:55 Panel Discussion

Session 3aMUa

Musical Acoustics: Acoustics of Percussion Instruments

Christopher Jasinski, Cochair
University of Notre Dame, 54162 Ironwood Road, South Bend, IN 46635

Andrew C. Morrison, Cochair
Natural Science Department, Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431

Invited Papers

9:00

3aMUa1. The harmonic vistaphone. Garry M. Kvistad (Woodstock Percussion, Inc., 167 DuBois Rd., Shokan, NY 12481, garry@chimes.com)

Percussion instruments offer a wide range of frequencies from the lowest gongs to the highest bells. The natural modes of vibrations of all percussion instruments produce overtones, which are inharmonic. Builders of keyboard percussion instruments must carve bars in specific ways to include a few harmonic overtones of their choosing. Garry Kvistad, founder/owner of Woodstock Percussion, Inc., has built a set of tubes/rods, which he calls the Vistaphone, carefully tuned to the first 32 partials of the harmonic overtone series. In this presentation, Mr. Kvistad will demonstrate how polyrhythmic patterns played slowly can be sped up by means of a proprietary software program to yield musical intervals that correspond to the ratios of just intonation. After this introduction illustrating the relationship of pulse and pitch, the Vistaphone will be played to reinforce the phenomena of pure harmonic relationships. The listener will be able to experience the pure harmonic overtone series in just intonation which yields a rich bouquet of sustained sound with a multitude of harmonically related difference/summation tones. The only way to obtain this effect acoustically is to build a unique instrument of this type.

9:20

3aMUa2. Coupling of drumhead vibrations: Experimental and numerical analysis. Randy Worland and Benjamin Boe (Phys., Univ. of Puget Sound, 1500 N. Warner, Tacoma, WA 98416, worland@pugetsound.edu)

Many musical drums, such as snare drums, bass drums, and tom toms, consist of two stretched membranes at opposite ends of a cylindrical shell. Vibrations of the two heads are coupled acoustically by the enclosed air and mechanically by the shell itself. The degree of coupling varies with modal frequency and shape, and depends on several factors including the geometry of the drum shell (diameter and length) and the tensions of the two heads. Experimental measurements of coupled vibrations on drums of various sizes have been made using an electronic speckle-pattern interferometer to image the deflection shapes, amplitudes, and relative phases of both heads simultaneously. These measurements are compared with a finite element model of the system that illuminates the role of the vibrational patterns of the enclosed air in the coupling process.

9:40

3aMUa3. From laboratory to concert hall: The invention and production of a new drum tuning system. Rohan Krishnamurthy (Music, Ohlone College, 544 Sunrise Circle, San Francisco, CA 94105, rohan.krishnamurthy@rochester.edu)

I discuss the motivations behind the invention and production of my new drumhead tuning system. The design originated with the *mridangam*, an ancient and popular pitched drum from South India, which has been constructed the same way for centuries. The mridangam is notoriously difficult to tune and replace drumheads, which consists of multiple layers of leather and a complex iron oxide and starch loading. My new design makes tuning and head replacement more user-friendly and durable, and can be applied to any drum that needs to be fine-tuned. I present my research comparing the acoustics of the traditional mridangam design with my new design. I also explore the challenges and opportunities involved in the international manufacturing process that is currently underway. The presentation concludes with an interactive performance on the mridangam.

10:00

3aMUa4. A method for predicting the performance of a steelband orchestra in arbitrary environments. David Chow and Brian Copeland (Dept. of Elec. and Comput. Eng., The Univ. of the West Indies, St. Augustine, Trinidad and Tobago, davidachow@gmail.com)

One of the challenges in steelpan performance is in determining the best placement of the various instruments in an orchestra so as to achieve the level performance desired by the musical arranger. This research attempts to develop a method for analyzing the performance of the tenor steelpan in arbitrary acoustic spaces using the room acoustic modeling software, Enhanced Acoustic Simulation for Engineers (EASE). When provided with an acoustic characterization of a speaker system, particularly the magnitude and phase radiation data, and the room in which it is located, EASE can be used to predict the resulting acoustic sound-field. EASE also requires an electrical equivalent of the instrument's sound source and, through its auralization feature, allow the user to actually hear an actual performance. EASE uses a variety of reconstruction techniques, such as ray tracing and boundary element method, to provide the most accurate responses. The study will verify the approach by comparing simulation results on direct sound coverage, total SPL, Clarity (C80), Centre Time, Reverberation, and Early decay times of the instrument with real time measurements in the usual 1/3-band octave frequency resolution. Although the work focuses on the tenor steelpan, it can be easily extended to predict the performance of an entire ensemble of steelpans in any given space.

Contributed Paper

10:20

3aMUa5. Ancient Aztec drum, the Huehuetl. Sergio Beristain (IMA, ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

The Aztecs and other cultures in Mesoamerica used to build and play during their festivities and warriors gatherings, simple drums made out from a hollow tree trunk, carved in the inside in order to make a fairly uniform thick surface to form the resonant chamber, topped by an ocelot skin, and open in the bottom, usually by three supports made from the same tree piece, called huehuetl, which means that these drums were of different shapes and sizes, some of these trunks were highly decorated by carving in the surface three dimensional images related to animals, faces, warriors, and other figures. These drums were excited on the top either by bare hand or wood sticks, varying the sound by the strength of the hit, and the area of the ocelot skin actually hit. Here are presented some pictures of ancient huehuetls together with the acoustic analysis of the sounds produced with a small replica of this percussion instrument.

WEDNESDAY MORNING, 20 MAY 2015 — BALLROOM 1, 11:00 A.M. TO 12:00 NOON

Session 3aMUb

Musical Acoustics: Acoustics of Percussion Instruments II: Concert

Garry Kvistad and Rohan Krishnamurthy will be demonstrating in concert the acoustics of various percussion instruments. All are welcome to attend this performance.

WEDNESDAY MORNING, 20 MAY 2015 — KINGS 5, 8:00 A.M. TO 11:40 A.M.

Session 3aNS

Noise, Architectural Acoustics, and ASA Committee on Standards: Noise with Tonal Components in the Built Environment

Lily M. Wang, Chair
Durham School of Architectural Engineering and Construction, University of Nebraska - Lincoln, PKI 101A, 1110 S. 67th St., Omaha, NE 68182-0816

Chair's Introduction—8:00

Invited Papers

8:05

3aNS1. An overview of issues in quantification of the tonalness of sounds. Patricia Davies (Ray W. Herrick Labs., School of Mech. Eng., Purdue Univ., 177 South Russell St., West Lafayette, IN 47907-2099, daviesp@purdue.edu)

Audible pitches in noise increase annoyance, and tone corrections to noise level metrics have been used in noise impact quantification for over 50 years. These corrections are a function of the prominence of the tone relative to the nearby spectral components. Frequency masking, the width of the tonal feature, and the frequency may also be taken into account. All measures are derived from spectral estimates. When a single or well isolated tonal components are present in noise, sounds are stationary and sufficient data is available for good spectral estimation, many tonalness metrics (Prominence Ratio, Tone to Noise Ratio, Tone Audibility, and Aures Tonality) work well. More challenging is quantification of tonalness when: the frequency is varying with time, either slowly or quickly; there is a harmonic series that is predominantly heard holistically not as a series of individual pitches; or there is beating because of the presence of multiple tones within a critical band. In music, the characteristics of harmonic families help us identify different instruments and manipulation of harmonic series in machinery noise can make tonal sounds more or less pleasant. The challenges of quantifying tonal contributions and their impact in these more complicated cases will be described

8:25

3aNS2. Tonal noise characteristics from HVAC equipment and how to improve practically and technically. Jyuhn-Jier J. Wang (Ingersoll-Rand PLC, 3600 Pammel Creek Rd., Bldg. 12-1, LaCrosse, WI 54601, jjwang@trane.com)

All HVAC equipment has rotating components, such as compressors and fans. These critical components often exhibit annoying tonal characteristics. Although there are many engineering methods to improve the tonal characteristics, there exists no commonly accepted tonal metrics in HVAC equipment specifications, which typically only include sound power or sound pressure levels at few operating conditions. However, the equipment sound level can vary greatly by the operating conditions—the range may be as much as 10 dB or more, depending on compressor/fan types, sizes, and speed range. Although HVAC manufacturers understand the impact of the equipment tonal characteristics to the acoustical comfort of built environment, lack of customer requirement in the annoyance of the tonal characteristics makes it very challenging to be assessed and improved upon with the other competing functional requirements. This presentation will show typical tonal noise characteristics of chillers, air handlers, and transport HVAC equipment, and then propose methods to improve practically and technically from equipment manufacturer's point of view.

8:45

3aNS3. Tonal noise in heating, ventilation, and air-conditioning equipment in commercial and residential buildings. Curtis Eichelberger, James E. Bender, Paul F. Bauch, and Jarom H. Giraud (Bldg. Efficiency, Johnson Controls, 631 Richland Ave., York, PA 17403, curtis.eichelberger@jci.com)

All heating, ventilation, and air-conditioning (HVAC) equipment that moves air or compresses refrigerant produces noise with tonal content. The challenge for the equipment manufacturer and building designer is to make sure that this tonal noise is isolated from the occupied spaces in a building. This paper provides an overview of the tonal content in airborne and structureborne sound produced by heating, cooling, and ventilation equipment. Examples of the tonal noise in HVAC components and packaged equipment will be presented. The challenge to manufacturers in measuring and rating equipment sound will also be covered. An overview of methods for testing and rating HVAC equipment sound will be presented, with emphasis on the uncertainty in characterizing the tonal content.

9:05

3aNS4. Transformers in the built environment. Felicia Doggett (Metropolitan Acoust., LLC, 40 W. Evergreen Ave., Ste. 108, Philadelphia, PA 19118, f.doggett@metro-acoustics.com)

Tonal noise in the built environment can be distracting, annoying, and at times make it practically impossible to live or work in a building. One of the worst offenders of tonal noise in buildings is electrical transformers. Whereas the airborne sound output of transformers is not particularly high, the vibration and subsequent structure-borne noise can be extreme. In large commercial buildings, transformers are typically installed directly on structural concrete floors. When these same transformers are located on upper floors of buildings, structure-borne sound transmission can be significant on the occupied floors below. For the most part, the transmission loss of the concrete floors is more than sufficient in attenuating airborne sound but structureborne sound can carry throughout the floors below for large distances from the transformer location. The best way to truly solve the problem is by isolating the transformer. So, how easy is it to elevate and isolate transformers that weigh as much as a truck? This presentation focuses on case studies of transformer noise in buildings that are incorrectly installed, what was subsequently done in an effort to remediate the problems, and whether or not it was successful.

9:25

3aNS5. Rumble strip sound level evaluation—Comparison of three designs. David Braslau (David Braslau Assoc., Inc., 6603 Queen Ave. S, Ste. N, Richfield, MN 55423, david@braslau.com) and Edward Terhaar (Wenck Assoc., Inc. , Maple Plain, MN)

Results of sound level monitoring from three types of longitudinal rumble strips installed along the edge of two-lane rural roads are discussed. This study was in response to objections by landowners about unwanted noise caused by vehicles traveling over rumble strips. The ultimate study goal is to provide maximum tactile and tonal sound levels at the driver position while minimizing exterior sound levels. Both exterior and vehicle interior sound levels were measured for three longitudinal edge-of-pavement rumble strip designs—California, Pennsylvania, and Minnesota. Simultaneous digital audio files were also recorded. Three vehicles types were tested at 30, 45, and 60 mph. Comparison of exterior levels and interior sound levels showed that the Pennsylvania design is the quietest, both interior and exterior. Interior levels from the Minnesota and California designs are similar but exterior levels are higher for the Minnesota design. The California design generates two tonal peaks. The Minnesota design generates only one higher tonal peak. Studies to date have not accurately addressed the distance at which a rumble strip signal can be heard. Sound levels were projected perpendicular to the roadway suggesting detectability at over a mile from the Minnesota design in a typical rural environment.

9:45–10:00 Break

10:00

3aNS6. Subjective evaluation of loudness and preference for noise containing audible tones. Stephan Toepken, Steven van de Par, and Reinhard Weber (Acoust. Group, Oldenburg Univ., Carl-von-Ossietzky-Str.9-11, Oldenburg 26129, Germany, stephan.toepken@uni-oldenburg.de)

Noise in living environments often contains unwanted tonal components that contribute to the unpleasant and intrusive character of a sound and usually lead to an increased annoyance. In the measurement and assessment of noise immissions after the German DIN 45681:2005-3 standard, the higher annoyance due to audible tonal components is covered by so called tone adjustments. Depending on the SNR of a tonal component within each critical band, the level of a sound can be charged with a penalty of up to 6 dB. In this study, the psychoacoustic penalty levels for sounds containing tonal components are directly determined in listening tests. The points of subjective equality (PSEs) for loudness and preference are determined using a matching procedure. The level of the test sound with tonal components is varied until it is equally loud / preferred as a tone-free reference sound that is constant in level. The level difference between test and reference sound at the PSEs is a direct measure of the penalty level. In listening tests with pure multi-tone sounds the penalty levels were found to be well above 6 dB, indicating that for these stimuli the German DIN standard is underestimating the penalty level.

10:20

3aNS7. Multidimensional characteristics of annoyance perception to tonal building mechanical noises. Joonhee Lee and Lily M. Wang (Durham School of Architectural Eng., Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, joonhee.lee@huskers.unl.edu)

Tones in noises from building mechanical systems can be a leading cause of complaints in an indoor environment. One of the greatest challenges to quantifying annoyance perception from these noises is the lack of information on how assorted acoustic characteristics relate to perceptual annoyance. Recent evidence suggests that annoyance perception by noises should be treated as a multidimensional rather than a unidimensional problem. Thus, the aim of this study is to explore multidimensional aspects of annoyance perception specific for building mechanical noises with tonal components. Tone frequency, tonal strength, presence of harmonics, and time fluctuation characteristics are investigated using actual noise recordings as well as artificially synthesized signals. Two subjective tests were implemented with the noise signals. Part A of the experiment is designed to identify prevailing acoustic characteristics for annoyance perception. The dominant acoustic characteristics were determined by multidimensional scaling analysis technique. A multidimensional annoyance model is subsequently proposed based on the test results. Part B is conducted to specifically investigate perceptual weighting of individual tones to overall contributions towards annoyance perceptions when complex tones are present in signals. The results of this test help to increase the accuracy of the developed annoyance model.

10:40

3aNS8. Psychoacoustically based tonality model for the evaluation of noise with tonal components. Roland Sottek (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, roland.sottek@head-acoustics.de)

For many years in various product assessments, tonality measurement procedures have been applied to identify and quantify prominent tonal components. The perception and evaluation of sound events containing such components has become increasingly important, e.g., in the field of vehicle acoustics for the assessment of tonality due to alternative drives, or in Information Technology (IT) devices due to hard disk drive noise. Additionally, many products include fans, e.g., IT devices, household appliances, and air-conditioning systems in buildings. These fans may emit prominent tonal sounds. The effective characterization of noise with tonal components is a challenge in acoustic and sound quality measurement. Existing methods for tonality calculation often show problems when applied to technical sounds where tonality perception is caused by different physical mechanisms like pure tones, multiple tones, narrowband noises, and even broadband noise showing very steep spectral slopes. To address this multitude of perceptual phenomena in a holistic approach, a new perceptually accurate tonality assessment method has been developed based on the hearing model of Sottek, evaluating the nonlinear and time-dependent specific loudness distributions of both tonal and broadband components via the autocorrelation function. The model has been validated by many listening tests. Its background and current state are presented.

Contributed Paper

11:00

3aNS9. Acoustic sensitivity phenomena found in people with cancer and thyroid problems. Bonnie Schnitta (SoundSense, LLC, 46 Newtown Ln., Ste. One, East Hampton, NY 11937, bonnie@soundsense.com)

Throughout my career in acoustic engineering, I have observed two interesting dynamics. Many of the people that I encountered with extreme noise sensitivity, that is easily disturbed by noise 1–2 dB above background (typically unperceivable by most), were found to be in ill health. Once I recognized this phenomena, I began to take note that there was an increase in these numbers specific to those suffering from cancer or a thyroid problem. I began to survey these individuals and most identified having been sensitive to noise long before their initial diagnosis, further developing my theory that noise sensitivity may indeed be a symptom for certain diseases, and a warning sign for early treatment. Second, those effected by ill health were specifically more bothered by lower frequency tones than those in good health. Many of these people are plagued by their noise sensitivity and describe it as a persistent nuisance that is often associated with inability to sleep with the tones present. This paper will discuss the knowledge of these facts, as well as a means to create a better healing environment in hospitals for patients that are noise sensitive.

11:15–11:40 Panel Discussion

Session 3aPP

Psychological and Physiological Acoustics: Spatial Hearing and Localization

Christopher Brown, Chair
University of Pittsburgh, 4033 Forbes Tower, 3600 Forbes at Atwood, Pittsburgh, PA 15260

Contributed Papers

8:00

3aPP1. Across-channel processing of interaural level differences. Matthew Goupell (Hearing and Speech Sci., Univ. of Maryland, 0119E Lefrak Hall, College Park, MD 20742, goupell@umd.edu)

Interaural level differences (ILDs) are produced when sounds occur away from the midline. The attenuation at the ear that is farther from the source location is frequency dependent and sometimes non-monotonic, resulting in inconsistent ILDs across frequency. It is unclear how the binaural system processes inconsistent ILDs. Ten normal-hearing listeners were presented one or three narrowband noises that varied in center frequency. Consistent or inconsistent ILDs were applied to the bands and listeners were asked to perform a left-right discrimination task. Diotic level roving was introduced to diminish listeners' ability to utilize monaural cues. Results show that thresholds were nearly frequency independent for one noise band. For three bands, thresholds became frequency dependent with the worst performance at 4 kHz, suggesting across-frequency ILD processing is a complex operation. These results have implications for people who do not heavily rely on low-frequency interaural time differences to localize sounds, such as bilateral cochlear-implant users.

8:15

3aPP2. Reverberation increases perceptual calibration to reliable spectral peaks in speech. Christian Stilp, Paul W. Anderson, Ashley Assgari, Gregory Ellis, and Pavel Zahorik (Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu)

Preceding acoustic context influences speech perception, especially when it features a reliable spectral property (i.e., relatively stable or recurring across time). When preceding sounds have a spectral peak matching F_2 of the following target vowel, listeners decrease reliance on F_2 and increase reliance on changing, more informative cues for vowel identification. This process, known as auditory perceptual calibration, has only been studied in anechoic conditions. Room reverberation smears spectral information across time, presumably giving listeners additional "looks" at reliable spectral peaks, which should increase the degree of perceptual calibration. Listeners identified vowels (varying from [i] to [u] in F_2 and spectral tilt) presented in isolation, then following a sentence filtered to share a spectral peak with the target vowel's F_2. Calibration was measured as the change in perceptual weights (standardized logistic regression coefficients) across blocks. Listeners completed sessions where stimuli were diotic or processed by virtual room acoustics to simulate high reverberation (T_{60} = 2.97 s). Perceptual calibration was significantly greater (larger decreases in F_2 weight, larger increases in tilt weight) for reverberant stimuli than diotic stimuli. Interestingly, hearing reverberant stimuli first attenuated calibration in subsequent diotic stimuli. Results demonstrate perceptual sensitivity to the persistence of reliable spectral peaks across time.

8:30

3aPP3. The effect of manipulating interaural level differences on lateralization by bilateral cochlear implant users. Christopher A. Brown (Univ. of Pittsburgh, 4033 Forbes Tower, 3600 Forbes at Atwood, Pittsburgh, PA 15260, cbrown1@pitt.edu)

Bilateral cochlear implant (BCI) users are relatively sensitive to interaural level differences (ILDs), but have been shown to be less sensitive to interaural time differences (ITDs). A proposed method for improving spatial release from masking for BCI users was recently demonstrated to do so, but was tested in only one spatial configuration: when one sound source was to the left, and the other to the right (Brown, Ear Hear. 2014). Briefly, instantaneous ITDs were measured and converted to ILDs using a linear function in which the applied ILD increased linearly with increases in the estimated ITD. Follow-up lateralization data showed that the effect of the proposed method was to hyper-lateralize sound sources, which proved effective in the configuration originally tested but may be less effective, or even deleterious, in other spatial configurations. In the current study, exponential functions are used, in which the applied ILDs are small when the ITDs are small, and increase more rapidly as the ITD moves away from midline. Results show significantly reduced RMS error in a lateralization paradigm with exponential functions.

8:45

3aPP4. Front-back confusions when sources and listeners rotate. William Yost and Xuan Zhong (Speech and Hearing Sci., ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

Tonal stimuli presented from directly in-front or directly in-back of a stationary listener are often miss-located in that sounds presented from in-front are occasionally perceived as being in-back of the listener; and often sounds presented in-back of a listener are perceived as being in-front. These types of front-back/back-front confusions in the azimuth plane were investigated as a function of the type of stimuli (e.g., tones, speech, and wideband noise) used and as a function of the sound or listener rotating around an azimuthal 24-loudspeaker array. Front-back confusions/back-front confusions depended on the type of stimulus; and on the number of loudspeaker positions the sound rotated through and the rate of sound or listener rotation. For example, if the sound rotated through two loudspeakers from in-front to in-back, there were a large number of confusions; if the sound rotated through each of the 24 loudspeakers on the circular array there were almost no confusions. [Research supported by an AFOSR grant.]

9:00

3aPP5. Binaural masking-patterns with short signals. Bjoern Luebken (Dept. of Experimental Audiol., Otto-von-Guericke Univ. Magdeburg, Leipziger Str. 44, Magdeburg 39120, Germany, bjoern.luebken@med.ovgu.de), Ifat Yasin (UCL Ear Inst., London, United Kingdom), G B. Henning (Colour and Vision Group, The Inst. of Ophthalmology, London, United Kingdom), and Jesko L. Verhey (Dept. of Experimental Audiol., Otto-von-Guericke Univ. Magdeburg, Magdeburg, Germany)

In masking-pattern experiments, thresholds are commonly measured for a sinusoidal signal masked by a narrow-band noise as a function of the

signal frequency. For a given spectral separation of signal and masker, beating between signal and masker may provide temporal fluctuation cues in addition to energy cues to detect the signal. In a dichotic condition, significantly broader masking patterns were found. It was hypothesized that this is due to the fact that the modulation cues do not play a major role in binaural processing. The present study investigates how masking patterns depend on signal duration. Masking patterns were not only measured for long (600 ms) signals but also for short signals (12 ms), where modulation cues should hardly play a role in signal detection even for monaural detection. The results show broader masking patterns for short signals than for long signals. In addition the binaural benefit does not change as much for the short signals than for the long signals when the signal frequency is varied. A binaural equalization cancellation model predicts the duration dependence of the masking patterns when a modulation analysis is assumed for the monaural pathway only.

9:15

3aPP6. When does binaural hearing benefit from ongoing envelope fluctuations? G. Christopher Stecker and Anna C. Diedesch (Hearing and Speech Sci., Vanderbilt Univ. School of Medicine, 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232-8242, g.christopher.stecker@vanderbilt.edu)

A number of classic and recent studies suggest that listeners' access to ongoing binaural information in sounds with steady envelopes is significantly poorer than for sounds with stochastic or slowly modulated ongoing envelopes. For some types of binaural cues—e.g., envelope interaural time differences (ITD)—that result is consistent with the requirement of salient envelope features for accurate cue encoding. For interaural level differences (ILD) and fine-structure ITD, however, no such requirement exists and the result is therefore more surprising. Regardless, the overall results suggest that binaural-cue encoding-in general-relies heavily upon the occurrence of infrequent envelope fluctuations, such as sound onsets and slow (e.g., syllabic-rate) amplitude modulations. Here, the results of relevant studies are reviewed, along with conceptual and physiological models of sampling binaural cues during envelope fluctuations. Potential advantages for binaural hearing in reverberation and competing backgrounds are considered, as is the impact of such mechanisms on the localization of strongly modulated targets such as speech. Overall results suggest a general degradation of binaural information-regardless of cue type-in the absence of envelope fluctuations. Instead, the binaural system emphasizes representations of temporally sparse, but highly localizable, onset-like events in the auditory scene. [Work supported by NIH R01-DC011548.]

9:30

3aPP7. Auditory localization performance in the azimuth for tactical communication and protection systems. Jeremy R. Gaston, Timothy Mermagen, Ashley Foots, and Kelly Dickerson (Dept. of the Army, US Army Res. Lab., Attn: RDRL-HRS, Aberdeen Proving Ground, MD 21005, jgaston2@gmail.com)

An important consideration in assessing auditory abilities is the effect personal protective equipment (PPE) can have on performance. Tactical Communication and Protection (TCAP) systems are a type of PPE that is becoming more common in military applications. These types of systems provide radio communications, while also protecting the user from hazardous noise through passive and active attenuation. These systems have an active acoustic pass-through mechanism that utilizes an external microphone and internal earphone to restore environmental hearing. Past research has shown that this type of pass-through mechanism can negatively affect localization ability. In the present study, six participants localization ability was tested for two different signals (AK-47 impulse, 500 ms white noise) across an array of eight loudspeakers arranged in a ring around the test participant. The systems tested consisted of three types of in-ear TCAPs, one over-the-ear TCAPs, and one passive nonlinear hearing protection device. All systems were tested with the Advanced Combat Helmet (ACH) donned. In general, localization performance was worst for the AK-47 impulse signal. Across signals, the over-the-ear TCAPs had significantly more front/back confusions and greater localization error than the in-ear systems.

9:45–10:00 Break

10:00

3aPP8. Perceptual effects of update-delay in room auralization. Samuel W. Clapp and Bernhard U. Seeber (Audio Information Processing, Technische Universität München, Arcisstr. 21, München 80333, Germany, samuel.clapp@tum.de)

A real-time auralization system, permitting sources and receivers to move about a simulated acoustic space, offers the opportunity to study auditory perception in realistic listening scenarios. However, such a system requires substantial computing resources to achieve the low latency needed for the simulation to adjust to new conditions without introducing artifacts. To use computing resources most efficiently from a perceptual standpoint, the perceptual effects of the simulation update delays must be understood. Simulated early reflections are typically determined via the image source method. When the source or receiver moves, the positions of the image sources relative to the receiver must be updated, with higher orders requiring longer calculation times. The aim of this study is to examine the perception of sound sources in simulated reverberant rooms when a source moves to a new position with only a partial update of image source locations. The parameters tested are perceived localization and apparent source width, as a function of the highest order of image sources updated and the distance traveled by the moving source. The results are meant to aid designers of real-time auralization systems by providing a perceptual basis for the necessary update rates for image source locations. They also aim to provide insight into the perception of auditory images that are incongruous with a real room, where there is a dissociation between source and reflection positions. [Funded by BMBF 01 GQ 1004B.]

10:15

3aPP9. The influence of signal type on the internal representation of a room in the auditory system. Elizabeth Teret, Jonas Braasch, and M. Torben Pastore (Rensselaer Polytechnic Inst., School of Architecture (Architectural Acoustics), Rensselaer Polytechnic Inst., 110 8th St. - Greene Bldg., Troy, NY 12180, terete@rpi.edu)

Currently, architectural acousticians make no real distinction between a room impulse response and the auditory system's internal representation of a room. With this lack of a good model for the auditory representation of a room, it is indirectly assumed that our internal representation of a room is independent of the sound source needed to make the room characteristics audible. In a perceptual test, we investigate the extent to which this assumption holds true. Listeners are presented with various pairs of signals (music, speech, and noise) convolved with impulse responses for different rooms. They are asked to evaluate the differences between rooms and disregard differences between the source signals. Multidimensional scaling is used to determine the extent to which the source signal influences the internal representation of the room and which room acoustical characteristics are important/perceivable for each sound type.

10:30

3aPP10. Rate effects in interaural and sequential level difference perception. Bernhard Laback (Acoust. Res. Institute, Austrian Acad. of Sci., Wohllebengasse 12-14, Vienna 1040, Austria, bernhard.laback@oeaw.ac.at), Mathias Dietz, and Stephan Ewert (Medizinische Physik, Universität Oldenburg, Oldenburg, Germany)

The relative weighting of post-onset interaural level difference (ILD) cues decreases with increasing modulation rate of a signal. It is unclear, however, whether overall ILD sensitivity decreases with increasing rate, particularly if stimulus duration and loudness are kept constant. Moreover, it is unclear if the rate effect arises also in monaural sequential level discrimination. In experiment 1, ILD-based lateralization discrimination thresholds and sequential level difference (SLD) discrimination thresholds were measured using bandpass-filtered pulse trains (4 kHz) with rates of 100, 400, and 800 pulses/s. From 100 to 400 pulses/s, both ILD and SLD thresholds decreased, contrary to a rate limitation. From 400 to 800 pulses/s ILD thresholds increased while SLD thresholds remained constant. The latter result is consistent with a high-rate limitation being specific to binaural hearing. Experiment 2 evaluated whether this rate limitation is due to the loss of transmitted modulation at high rates. The ILD thresholds for an unmodulated 4-kHz pure tone were indeed higher than those for pulse-

trains. An auditory nerve model (Zilany *et al.*, 2014, JASA **135**, 283–286) combined with an interaural discharge rate comparison stage qualitatively predicted the nonmonotonic pattern of ILD thresholds. Overall, the results suggest that modulation can be beneficial for ILD perception.

10:45

3aPP11. Just noticeable differences of listener envelopment and apparent source width. Stefan Klockgether and Steven van de Par (Medical Phys. and Acoust., Carl von Ossietzky Univ., Carl-von-Ossietzky-Str. 9-11, Oldenburg, Lower Saxony 26129, Germany, stefan.klockgether@uni-oldenburg.de)

Listener envelopment (LEV) and apparent source width (ASW) are attributes which describe the spatial perception of a sound in a room. They are linked to several parameters of physically measurable binaural room impulse responses (BRIR), e.g., the interaural cross-correlation (IACC). Both the LEV and ASW decrease with increasing IACC. For this study the IACC of the BRIR of different rooms was directly manipulated to systematically change the spatial impression of a sound in a room. The BRIRs were convolved with anechoic music signals, and the resulting stimuli were presented to subjects in a psychoacoustic experiment to estimate the just noticeable differences of LEV and ASW. For that purpose, the subjects had to rate the different stimuli for both perceptual attributes in seven steps from "no envelopment" to "completely enveloped" or from "small" to "wide" respectively. In a second step, the stimuli which were rated the same, were used in a paired comparison paradigm to obtain a more precise estimation of the ability to distinguish between similar stimuli. The results of this study are used in a psychoacoustically motivated model to estimate differences in perceived spaciousness of sounds in rooms by the prediction of the perceived envelopment and source width.

11:00

3aPP12. A Bayesian framework for the estimation of head-related transfer functions. Griffin D. Romigh (Air Force Res. Labs, 2610 Seventh St., Area B, Bldg. 441, Wright Patterson AFB, OH 45433, griffin.romigh@us.af.mil), Richard M. Stern (Carnegie Mellon Univ., Pittsburgh, PA), Douglas S. Brungart (Walter Reed National Military Medical Ctr., Bethesda, MD), and Brian D. Simpson (Air Force Res. Labs, Dayton, OH)

While high-fidelity spatial auditory displays require individualized head-related transfer functions (HRTFs), much of the physical structure contained within an HRTF is similar across most individuals. This suggests that a Bayesian estimation technique, which uses sample observations to bias an a priori model towards an individual, may provide benefits in terms of efficiency. Therefore, the current work proposes a Bayesian HRTF framework that utilizes HRTFs in the form of their real spherical harmonic representation. When combined with assumptions of normality, the resulting technique is shown to enable the accurate estimation of an individualized HRTF from a small set of spatially distributed measurements. Moreover, the model provides a convenient way to analyze which components of an HRTF vary most across individuals, and can therefore be used to create very efficient HRTF measurement strategies. A perceptual localization test confirmed that similar localization performance could be attained with an HRTF estimated from as few as 12 spatial measurements, even when confined to the median-sagittal plane.

WEDNESDAY MORNING, 20 MAY 2015 KINGS 3, 8:00 A.M. TO 12:00 NOON

Session 3aSA

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

Kuangcheng Wu, Cochair
Ship Survivability, Newport News Shipbuild, 4101 Washington Ave., Newport News, VA 23693

Jeffrey Cipolla, Cochair
Weidlinger Associates, Inc, 1825 K St NW, Suite 350, Washington, DC 20006

Invited Papers

8:00

3aSA1. Acoustic radiation from an infinite submerged, line-driven plate with attached finite plate. 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)

The structural acoustics and vibration related to submerged, structurally driven elastic plates is germane to a wide range of applications in U.S. Navy undersea vehicle and system designs. Of particular interest to a recent specific Naval design application was the examination of effects on the acoustic radiation from a very large, fluid-loaded elastic plate joined with a "welded," or strongly attached, smaller elastic plate subjected to force-loading excitation somewhere on the large plate. This paper presents the results following an examination of several theoretical approaches to predicting the radiated sound level (both levels and directivity) of an idealized infinite plate and perfectly attached thin, finite plate submerged in a heavy fluid and driven by a line-load at varying locations on the infinite plate. Parametric studies are also presented to both highlight the underlying basic physical mechanisms and also to investigate the effect on sound radiation of varying system parameters. Specifically, results of parametric variations in (a) location of the line force on the plate with respect to the attached finite plate, (b) length/size of the attached plate, (c) mass and/or stiffness of the attached plate, (d) frequency of the harmonic line-load, and, time permitting, and (e) the mechanical theory of the attached thin plate, are all presented herein.

8:20

3aSA2. Causal and passive interpolation of structural acoustic matrices. James G. McDaniel and Andrew S. Wixom (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, jgm@bu.edu)

The present work investigates the causal and passive interpolation of structural acoustic matrices in the frequency domain. Interpolations significantly ease computational burdens involved in computing these matrices from models. For example, one may wish to compute a small admittance matrix, relating forces to velocities at points on a structure, from a large finite element model with over one million degrees of freedom. In many structural acoustic systems, the elements of such matrices vary smoothly with frequency and therefore interpolation can be accurate and effective. In the time domain, causality requires no effect before its cause. In the frequency domain, this takes the form of Hilbert transform relations that relate the real and imaginary parts of scalar transfer functions. Passivity requires that the net power dissipated by the structure over a cycle of vibration be positive semidefinite. The present work develops series expansions in which these conditions may be implicitly satisfied by the proper choice of basis functions. Coefficients of in the series expansion are found by matching the series to the known values at the interpolation points. Examples demonstrate situations in which the satisfaction of these physical conditions yields more accurate interpolations.

8:40

3aSA3. A hybrid method for predicting heavy fluid loading of structures. Micah R. Shepherd and John B. Fahnline (Appl. Res. Lab, Penn State Univ., PO Box 30, Mailstop 3220B, State College, PA 16801, mrs30@psu.edu)

Measuring the resonance frequencies and damping of structures in heavy fluids can be difficult and time consuming. To alleviate this burden, a hybrid approach is proposed by combining in-air experimental modal analysis with a boundary element estimate of the acoustic resistance and reactance matrices. The method was validated on a rectangular metal plate by comparing the natural frequencies estimated using the hybrid method to the natural frequencies measured in water. The modal parameters were first measured in air with the mass-normalized mode shapes estimated using the complex mode indicator function. The modal parameters were then updated using rational polynomials curve fitting. The measured modes were then used as *in-vacuo* basis functions to compute modal resistance and reactance matrices in water and the fluid-loaded vibration and sound radiation was estimated. The natural frequencies of the plate submerged in water were then measured and compared to the frequencies predicted by the hybrid method showing good agreement.

9:00

3aSA4. Jacobian-Free Newton Krylov based iterative strategies in frequency-domain analyses. Jeffrey Cipolla (Weidlinger Assoc., Inc, 1825 K St NW, Ste. 350, Washington, DC 20006, cipolla@wai.com) and Abilash Nair (Weidlinger Assoc., Inc, New York, NY)

This study examines the theory and performance of Jacobian-Free Newton Krylov (JFNK) methods for the efficient, iterative solution of steady state dynamics of linear vibrating systems in the frequency domain. Currently, most commercial FEM algorithms employ use direct factorization of large system matrices to achieve steady state solution. Such approaches are usually quite demanding on memory and CPU requirements. Some implementations exploit iterative solutions, but still use large, assembled system matrices, requiring significant memory. The methods investigated in this work avoid the formation of a matrix completely, minimizing memory requirements and enabling much larger problems to be performed on desktop computers. Here, the Conjugate-Gradient (CG) and Transpose Free Quasi Minimal Residual (TFQMR) algorithms are studied as possibilities. These methods are of particular interest for adaptation to finite element software which uses explicit transient dynamics, because such software's optimal architecture prevents the formation and solution of a stiffness (or a tangent stiffness) matrix. In order to demonstrate the advantages of these algorithms, we choose examples that use the JFNK-CG and JFNK-TFQMR technique to show computational advantages over regular matrix based solutions, with significant decrease in memory requirements.

Contributed Papers

9:20

3aSA5. Random sound pressure fluctuations Induced by non-linear damping in a reactor column. Hasson M. Tavossi (Phys., Astronomy, and GeoSci., Valdosta State Univ., 2402, Spring Valley Cir, Valdosta, GA 31602, htavossi@valdosta.edu)

Self-excited non-harmonic oscillations due to non-linear damping in the system are produced in a reactor column. The uniform steady flow is converted spontaneously into random fluctuations in sound pressure, under the especial experimental conditions, in a reactor column that includes a thin layer of dissipative porous medium. The resulting large-amplitude non-harmonic pressure fluctuations in the air-flow are similar to the bifurcation in chaotic systems; where two or more energy states can occur simultaneously, and the system oscillates between them. Experimental results demonstrate this abrupt change in flow-regime, from steady-flow to random fluctuations. Our results show that a low-pressure shock-wave- is established that precedes the self-excitation fluctuation in the system. Results also show the existence of a threshold for flow-velocity beyond which this transition from steady to pulsating non-harmonic pressure fluctuations occur. A numerical model is developed for this behavior in terms of; FFT peak frequencies, flow-velocity, characteristic numbers, relaxation-time, and system nonlinear damping.

9:35

3aSA6. Numerical hybrid TMM-modal FEM method prediction of the vibroacoustic of sandwich panels with add-on damping. Imen Rzig and Noureddine Atalla (mécanique, univesité de Sherbrooke, 2500 Boulevard Univ., Sherbrooke, Quebec J1K2R1, Canada, imen.rzig@usherbrooke.ca)

This paper discusses the numerical modeling of the vibroacoustic response of sandwich-composite panels with add-on damping, under acoustic excitation, diffuse acoustic field (DAF). A modal synthesis approach is used for the calculation of the structural response and the Rayleigh's integral is used for the acoustic response. Since the panel has a viscoelastic core, a methodology is presented to handle efficiently the modeling of the frequency depended properties of the viscoelastic layer. A hybrid TMM-modal FEM method is used to predict the acoustic response in high frequency, using the equivalents properties of panel, which are calculated from strain energies' panel. Next, a parameters' study on the effect of the viscoelastic layer location is presented. In particular, three locations are compared: within the Honeycomb core, within the skins and added to the skin with a constraining layer. The effects of the excitation type on the vibration and acoustic response are also discussed using the hybrid: TMM-modal FEM method. Key words: Sandwich NIDA, Modal FEM method, TMM method, viscoelastic damping, acoustic response, equivalent properties.

9:50–10:15 Break

10:15

3aSA7. Reconstructing excitation forces acting on a baffled plate using near-field acoustical holography. Pan Zhou (College of Power and Energy Eng., Harbin Eng. Univ., 5050 Anthony Wayne Dr., Wayne State Univ., Detroit, MI 48202, sean.f.wu@gmail.com), Sean F. Wu (Mech. Eng., Wayne State Univ., Detroit, MI), and Wanyou Li (College of Power and Energy Eng., Harbin Eng. Univ., Harbin, Heilongjiang, China)

This paper presents numerical simulations of using nearfield acoustical holography technology to reconstruct the excitation forces acting on the surface of a baffled plate. The reason for choosing a baffled plate is that analytic solutions to the vibration responses and acoustic radiation are readily available. To acquire a better understanding of the interrelationship between excitation forces and vibro-acoustic responses of the plate, the structural intensities and power flows due to individual excitation forces and their interactions are calculated. The relative contributions of the excitations toward overall structural vibration responses of the plate are analyzed by taking the normal modes transform into the wavenumber domain. Results show that the wavenumber contents of vibration responses of a plate due to a line, point, and distributed forces have very distinct characteristics. Therefore, by analyzing the vibration responses in the wavenumber domain, it is possible to reveal the impacts of individual excitation forces on the resultant structural vibration of a plate.

10:30

3aSA8. Magnetic excitation and identification of flexural modes of a circular plate. Timothy D. Daniel, Phil L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu), Ahmad T. Abawi (HLS Res., La Jolla, CA), and Ivars Kirsteins (NUWC, Newport, RI)

A spatially localized, oscillating, magnetic field was used to induce resonant vibrations in an aluminum circular plate without direct mechanical contact. Experiments were done with the plate in air, at a free surface, and fully submerged in water. This allowed the effect of fluid loading on the plate's spectrum to be examined for both the fully loaded and half loaded case. Since the magnetic field is spatially localized, identification of the mode shapes is possible by scanning the source along the target and measuring the varying response of the plate. Additional experiments were done with an open-ended fluid filled copper circular cylinder in which both resonant frequencies and mode shapes were identified. Both targets responded at twice the oscillation frequency of the applied field [B. T. Hefner and P. L. Marston, J. Acoust. Soc. Am. **106**, 3340–3347 (1999)]. The response corresponds to the oscillation frequency of the Maxwell stress associated with combined applied field and induced eddy-current field. Excitation of both plate and cylinder was found to be measurable with a steady state and a tone burst source. [Work supported by ONR.]

10:45

3aSA9. Localized generation of a vibration pattern on a thin plate by using traveling wave control with periphery actuator array. Jung-Han Woo and Jeong-Guon Ih (Mech. Eng., KAIST, Korea Adv. Inst. of Sci., Guseong-dong, Yuseong-gu, Daejeon 305701, South Korea, j.h.woo@kaist.ac.kr)

Vibrations are rendered on the display panel of a mobile electronic device to transfer a tactile sensation to users for effective information. Previous vibration control has usually focused on the suppression of an entire area. The only choice for actuator position is the boundary of the panel due to the utilization of central part for other electric components by consideration of the practical application. Traveling wave control is adopted as the basic principle to generate a rendered vibration pattern. Proper weighting for both amplitude and phase for each actuator in the array is determined by applying the general inverse method with the relationship velocity responses and input signals of actuators. Experiments are conducted on a thin the tempered glass panel of a commercial tablet PC employing small moving-coil actuators. Good performance to fulfill the rendered 2x2 and 3x3 grid distribution of vibration magnitude was observed with more than 97 % success ratio. Also, Regularization provides 75% reduction of the control effort. Minimum number and optimum position of actuators for rendering target vibration pattern can be chosen by consideration of the independency of actuator array.

11:00

3aSA10. Building leakage detection and quantification using statistically optimized nearfield acoustic holography technique. Kanthasamy Chelliah, Ganesh G. Raman (Illinois Inst. of Technol., 10 w 32nd St., Ste. 243, Chicago, IL 60616, kchellia@iit.edu), Ralph T. Muehleisen (Argonne National Lab., Argonne, IL), Hirenkumar Patel (Illinois Inst. of Technol., Chicago, IL), and Eric Tatara (Argonne National Lab., Argonne, IL)

This paper presents a building infiltration detection and quantification technique using statistically optimized nearfield acoustic holography (SONAH) technique. A model building with known cracks on its wall was investigated in this study. The model building housed a synthetic acoustic source. A hologram measurement was performed outside the model building and the sound pressure levels on the walls were reconstructed. The correlation between the reconstructed pressure levels and the area of the crack was obtained. It was found that the acoustic technique can successfully detect and quantify the leakages from the building model. Two different frequencies of reconstructions are compared and it was found that the lower frequency reconstructions were more accurate. Effects of various regularization methods, phase matching, and quantization error are discussed. Various methods to suppress the wrap-around error in the reconstructions will be addressed.

11:15

3aSA11. Vibrational analysis of hollow and foam-filled graphite tennis rackets. Kritika Vayur (Graduate Program in Acoust., Penn State Univ., 729 S. Allen St., State College, PA 16801, kuv126@psu.edu) and Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., University Park, PA)

The game of tennis is plagued with wrist and elbow injuries. Most players use light, stiff hollow graphite rackets. A recent foam-filled racket design appears to reduce the risk of wrist injury. A vibrational analysis of a wide variety of rackets was conducted in attempt to understand the science behind this racket's apparent success. Damping rates were measured for the bending and torsional modes for a wide variety of hollow and foam-filled rackets. No significant difference among the rackets was found, suggesting the benefit of the foam-filled racket design is not due to damping. An extensive modal analysis was conducted for several rackets over the frequency range of 100 Hz to 1.5 kHz to identify the bending, torsional, and string modes. In this paper, the damping rates, mode shapes, and frequencies, will be compared for several hollow and foam-filled rackets. Results will be interpreted in light of the relevant forces between the racket, ball, and arm.

11:30

3aSA12. Frequency eigenvalues of the breathing mode of a submerged empty thin spherical shell: Variation with the ratio of shell thickness to radius. Marshall V. Hall (Marshall Hall Acoust., 9 Moya Crescent, Kingsgrove, New South Wales 2208, Australia, marshallhall@optushome.com.au)

The frequency eigenvalues of the breathing mode of a submerged empty thin spherical shell are the three roots of a cubic algebraic equation. Compared with the quadratic equation for an *in-vacuo* shell, the polynomial contains two imaginary terms in the ratio of sound-speed in the ambient fluid to plate velocity of the shell material. All three roots of the equation are complex, due to radiation into the ambient fluid. For a shell submerged in a fluid similar to water, two of the eigenvalues' real parts, and two of their imaginary parts, do not vary monotonically with shell thickness. To illustrate this phenomenon, an example of a steel shell submerged in water is considered. For a thickness to radius ratio of 0.1, one of the non-dimensional eigenvalues has a real part of 1.57, close to the *in-vacuo* result of 1.61. As the thickness ratio decreases to 0.0143, this real part decreases to zero and remains there for lower ratios. Another eigenvalue, whose real part is always zero above 0.0143, has a positive real part at ratios less than 0.0141, with a maximum at 0.009. Over the interval from 0.0141 to 0.0143, however, the real parts of all three eigenvalues are zero.

11:45

3aSA13. Bias of vibrational energy at high frequencies in circular and rectangular plates. Thomas D. Boyer (Appl. Res. Lab, Penn State Univ., 200 GTWT, University Park, PA 16802, tub139@psu.edu) and Micah R. Shepherd (Appl. Res. Lab, Penn State Univ., State College, PA)

Experimental modal analysis can be used to measure the modal parameters and vibrational energy in structures. This presentation will compare biases in the total vibration energy at high frequencies for a circular and rectangular aluminum plate that are freely supported. Both rectangular and cylindrical coordinates were used to create the set of excitation points for the circular plate, while only rectangular coordinates were used for the rectangular plate. Singular value decomposition was then used to obtain the natural frequencies and mode shapes in each of these plates. The obtained modes were used to synthesize the vibration response and to compute the vibration energy. Biases were found at high frequency using a direct integration method to compute the vibration energy. The dependencies of the bias on the plate geometry and set of excitation points will be discussed.

WEDNESDAY MORNING, 20 MAY 2015 COMMONWEALTH 2, 8:00 A.M. TO 11:20 A.M.

Session 3aSC

Speech Communication: Celebration of Kenneth N. Stevens' Contributions to Speech Communication: Past, Present, Future I

Helen Hanson, Cochair
ECE Dept., Union College, 807 Union St., Schenectady, NY 12308

Stefanie Shattuck-Hufnagel, Cochair
MIT, 36-511 MIT, 77 Mass Ave., Cambridge, MA 02139

Chair's Introduction—8:00

Invited Papers

8:05

3aSC1. Ken Stevens and linguistic phonetics. Patricia Keating (Linguist, UCLA, Los Angeles, CA 90095-1543, keating@humnet.ucla.edu)

While almost all of Ken Stevens's research has been influential in linguistic phonetics—especially his work on the quantal nature of speech and on enhancement theory—this presentation will focus more on aspects not covered by others in this session. A noteworthy aspect of Ken's career is that he frequently collaborated with linguists, notably in research on phonetic features and their structure. In work with Blumstein and with Halle, he provided acoustic correlates of place of articulation, laryngeal, and nasalization features. Much of this work is summarized in his 1980 paper in *JASA*, "Acoustic correlates of some phonetic categories." In work with Keyser, he suggested an overall organization of features to define major classes of sounds. This work was published in linguistics journals, e.g., their 1994 paper in *Phonology*, "Feature geometry and the vocal tract." Ken's importance to linguistics, as an engineer interested in linguistic sound systems and eager to work with phoneticians and phonologists, cannot be overestimated, and is a legacy continued by many of his students.

8:20

3aSC2. Distinctive features, quantal theory, and enhancement. Helen Hanson (ECE Dept., Union College, 807 Union St., Schenectady, NY 12308, helen.hanson@alum.mit.edu)

Ken's background in physics and engineering and keen interest in phonetics gave him a rare ability to suggest a quantitative basis for distinctive features. His awareness that the acoustic correlate of a smoothly varying articulatory parameter might have a sudden change in value led to his development of Quantal Theory. In addition to providing a basis for distinctive contrasts, it provides a means for a continuous-time signal to reflect the discrete nature of the underlying representation of speech sounds. Later work led to two types of quantal relations: those based on aeromechanical properties of the vocal tract, which lead to manner features, and those based on coupling between cavities, which lead to articulator-bound features. Ken also suggested that each feature has a defining articulatory state, and thus, a defining acoustic correlate. Neighboring segments or higher-level phenomena may threaten the salience of defining acoustic correlates. Enhancement theory was developed to account for the fact that the defining articulatory or acoustic correlates are threatened in some environments. At such times, enhancing gestures that bolster the salience of the defining acoustic correlates are brought into play. Much of the work produced by Ken's students was aimed at finding support for these theories.

8:35

3aSC3. The role of subglottal resonances in speech processing algorithms. Abeer Alwan (Elec. Eng., UCLA, 405 Hilgard Ave., Los Angeles, CA 90095, alwan_99@yahoo.com), Steven Lulich (Speech and Hearing Sci., Indiana Univ., IN), and Harish Ariskere (Xerox-India, Bangalore, India)

Inspired by Ken Stevens' work on speech production, we investigated the role of the subglottal resonances (SGRs) in several machine-based speech processing algorithms. The subglottal acoustic system refers to the acoustic system below the glottis, which consists of the trachea, bronchi, and lungs. The work is based on the observations that the first and second subglottal resonances (Sg1 and Sg2) form phonological vowel feature boundaries, and that SGRs, especially Sg2, are fairly constant for a given speaker across phonetic contexts and languages. After collecting an acoustic and subglottal database of 50 adults and 48 children, analyzing the database, and developing algorithms to robustly estimate SGRs, we were able to use these resonances to improve the performance of a variety of speech processing algorithms including: recognition of children's speech in limited-data situations, frequency-warping for adult speech recognition, speaker recognition and speaker verification, and automatic height estimation. [Work supported in part by the NSF.]

8:50

3aSC4. Influences of Ken Stevens on research in speech production. Joseph S. Perkell (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Cambridge, MA 02445, perkell@mit.edu)

Ken Stevens has had a profound influence on many aspects of speech research. His contributions to the understanding of speech range from the theoretical through a vast body of hypothesis-driven studies to important developments in methodology. His theory of the quantal nature of speech has important implications for our understanding the nature of sound structures of languages. It embodies the concept that sounds are specified by combinations of distinctive features, which are based on universal properties of the production and perception mechanisms. Over a period of five decades, he, the many people he mentored and others tested hypotheses stemming from this theory using modeling and running experiments on relations among vocal-tract configurations, articulatory movements, tissue properties and aerodynamics—and the influences of these factors on perceptually relevant characteristics of the resulting speech sounds. Investigators who spent time in Stevens' Speech Communication Group also learned valuable principles for the conduct of scientific research. Examples of research on speech production will be presented to illustrate some of these points. [Research supported by the NIDCD, NIH.]

9:05

3aSC5. Ken Stevens' influence on aerodynamic models of fricatives. Christine H. Shadle (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu)

In the late 1950s, John Heinz, under Ken Stevens' supervision, used a constriction in a duct to model fricatives, recording the far-field sound spectra. Ken elaborated the model over the next 15 years: The constriction's location corresponded to place. The flowrate and constriction area were used to compute the Reynolds number, indicating the degree of turbulence of the flow; the pressure drop predicted the far-field sound. However, as I showed when Ken supervised my thesis in the 1980s, an obstacle downstream of the constriction created more turbulence noise, and was a better approximation to sibilants. It could still be modeled by a circuit analog with independent source and filter. Similar models of interdentals and palatals were not as successful; geometric details not included in the area function affected the acoustic output. Other limitations of early models include: circuit analogs use only the speed of sound, but the much-slower convection velocity plays a role in noise modulation in voiced fricatives; noise source changes during transitions are over-simplified; quasi-static modeling of fricatives excludes the noise produced by articulator movement. Ways of incorporating these mechanisms of fricative production in simple, conceptually powerful models are needed.

9:20

3aSC6. Ken Stevens' contributions to the field of communication disorders. Ray D. Kent (Waisman Ctr., Univ. of Wisconsin-Madison, Rm. 491, 1500 Highland Ave., Madison, WI 53705, kent@waisman.wisc.edu)

Speech is a robust faculty and most people rely on it daily, overcoming temporary difficulties such as laryngitis, dental work, or stage fright. Even slips of the tongue are relatively rare, occurring once in about a thousand spoken words. But for all its robustness, speech is vulnerable to a number of disorders that can be severe enough to interfere with efficient and natural communication. A communication disorder can be defined as a disruption or limitation in the production of speech that is fluent, intelligible, and natural. Through direct and indirect influences, Ken Stevens became a valued ally in the effort to bring relief to people with these disorders. In a highly selective account of the contributions of Ken and his colleagues, this talk examines three general areas: technological advances for the assessment and treatment of speech and voice disorders, basic theories that illuminated the path to understanding speech communication in health and disease, and quantitative research focused on speech communication in individuals with hearing loss or dysarthria (both of which have multisystem consequences on the production of speech and both of which can result in serious communication difficulties).

9:35

3aSC7. Looking into the future by looking back at the past: Observations about Ken Stevens' views on speech perception. David B. Pisoni (Psychol. and Brain Sci., Indiana Univ., 1101 E 10th St., Bloomington, IN 47405, pisoni@iupui.edu)

There has always been an interest in the perception of speech and non-speech signals. As Ken pointed out 35 years ago (Stevens, 1980), speech signals possess several highly distinctive acoustic properties that set them apart from other auditory signals. He argued that the study of the acoustic properties of classes of speech sounds that occur in natural languages may provide insights about the "special" response characteristics of the auditory system. He observed that all speech sounds have a small number of general properties in common: (1) they have a short-time spectrum that contains peaks and valleys; (2) they have up-and-down variations in amplitude as a function of time; and (3) the short-time spectrum changes over time. I summarize studies with normal-hearing and hearing-impaired listeners with cochlear implants that provide converging support for the conclusion that human listeners have developed highly efficient perceptual processing strategies to make optimal use of minimal acoustic information in the speech signal. [Work supported by grants from NIDCD to Indiana University.]

9:50–10:05 Break

10:05

3aSC8. Analysis by synthesis techniques. William Idsardi (Linguist, Univ. of Maryland, 1401 Marie Mount Hall, College Park, MD 20742, idsardi@umd.edu)

In the late 1950s and early 1960s, Ken Stevens and Morris Halle proposed *analysis by synthesis* (A-by-S) as a general model for speech perception by humans and machines. The leading idea in A-by-S is that people can generally both speak and listen, and therefore speech is both action and perception. Thus, the perception of a speech fragment can be the answer to the question, "how would I have said that?," that is, to recover the action that would produce that event (or an appropriately scaled version of the event). This idea has had lasting influence in speech perception models and systems (even if they employ other terms), as well as in more general accounts of perception. In this talk, we will offer a brief history of the idea of A-by-S, focusing on the general architecture of such systems and in what space (auditory or articulatory) signal comparison is carried out. In doing so, we will briefly compare and contrast A-by-S with other influential ideas, including the motor theory of speech perception, modularity and the special nature of speech, mirror neurons, the memory-action-perception loop, Bayesian inference, and general auditory cognition.

10:20

3aSC9. Ken Stevens, motor theory, and infant MEG brain data. Patricia K. Kuhl (Inst. for Learning & Brain Sci., Univ. of Washington, Box 357920, Seattle, WA 98195, pkkuhl@u.washington.edu)

ASA meetings provided a special time to talk to Ken Stevens and share data and opinions about topics of mutual interest. On one of these occasions, Ken uttered a sentence that stuck with me, "I hate to tell you this, but I think I'm becoming a motor theorist!" He then explained *Analysis by Synthesis*. In this talk, I'll share newly published MEG brain data (Kuhl *et al.*, *PNAS*, 2014), suggesting that young infants may indeed be engaging in something like analysis by synthesis as they listen to us speak. I'll use this and other findings to discuss Ken's important contributions to Speech Communication.

10:35

3aSC10. The MIT Lexical Access Project: A window into Stevens' model of feature-cue-based processing. Jeung-Yoon Choi and Stefanie Shattuck-Hufnagel (MIT, 50 Vassar St., Rm. 36-523, Cambridge, MA 02139, sshuf@mit.edu)

In his JASA (2002) paper, Ken Stevens proposed a model of human speech recognition based on extracting acoustic cues to the distinctive feature contrasts of the speaker's intended words. Following Halle (1995), he distinguished between cues to manner features (e.g., abrupt spectral events associated with constrictions/widenings of the vocal tract, called landmarks) vs. cues to other features related to voicing and place, and proposed that landmark detection is an early step in perception. Landmark cues are particularly useful: they are reliably produced, robustly detectable, and highly informative about the structure and lexical content of an utterance; they also identify adjacent regions rich in cues to additional features. During his working life, Ken's students developed many of the modules required to detect feature cues, meanwhile discovering important aspects of their systematic context-governed variation. Current work aims at (1) completing the speech analysis system based on detection of feature cues and parameter values, (2) evaluating its performance, and (3) comparing its performance to human perceptual behavior. A system based on Ken's insights will have implications for human speech processing models, for knowledge-based approaches to ASR, and for a deeper understanding of the mechanisms underlying clinical speech problems as well as language learning.

10:50

3aSC11. Ken Steven's research and influence on automatic speech recognition. Carol Y. Espy-Wilson (Elec. and Comput. Eng., Univ. of Maryland, A.V. Williams Bldg., College Park, MD 20742, espy@umd.edu)

This talk will address Dr. Kenneth Steven's considerable influence in the area of speech recognition. Central to Ken's model of lexical access is the relationship between acoustics and articulation. Binary distinctive features are identified from the continuous signal based on acoustic cues that are tied to articulation. As such, variability in this parameterization is reduced from that seen in the physical signal. The acoustic features are then matched against the lexicon where there is generally only one representation for each word. This lexical representation consists of a bundle of distinctive features. In contrast, state-of-the-art recognizers use acoustic features that are not tied to articulation. Consequently, they are highly variable, resulting in the recognizer's over-reliance on sophisticated machine-learning tools and statistical signal processing methodologies that are data intensive and often fail to generalize. Further, multiple lexical representations of each word in terms of acoustic units called tri-phones are needed to cope with pronunciation variability. While Ken's model has not been fully realized, it has motivated research into extraction of the distinctive features, investigations of alternate articulation-based representations, and marrying these representations with well-established recognition back-ends to study gains in robustness. Examples from studies and their promise relative to state-of-the-art methods will be given.

11:05

3aSC12. Ken Stevens' influence on the development of paradigms for speech synthesis. Brad H. Story (Speech, Lang., and Hearing Sci., Univ. of Arizona, 1131 E. 2nd St., P.O. Box 210071, Tucson, AZ 85721, bstory@email.arizona.edu)

Synthetic speech has long been used as a means of understanding both speech production and speech perception, as well as for technological applications such as text-to-speech devices. Paradigms for developing speech synthesis systems have included electrical circuits, digital filters, and computational models that replicate either the structure or acoustic characteristics of the voice source and vocal tract. This presentation will focus on how Ken Stevens' investigations of speech, spanning more than five decades, have directly influenced essentially every paradigm of speech synthesis, including formant synthesis, articulatory synthesis, and speech production modeling. [Work supported by NIH R01-DC011275 and NSF BCS-1145011.]

Session 3aSPa

Signal Processing in Acoustics: Smartphone Acoustic Signal Processing Student Competition (Poster Session)

Kevin Cockrell, Cochair
Applied Physical Sciences Corp., 2488 Historic Decatur Rd., Ste. 100, San Diego, CA 92106

Joshua Atkins, Cochair
Beats Electronics, LLC, 1431 Ocean Ave., Apt. 1018, Santa Monica, CA 90401

The ASA Technical Committee on Signal Processing in Acoustics is sponsoring a competition for students to develop concepts for smartphone applications ("apps") that make novel use acoustic signal processing. This session will include a poster describing each smartphone app concept. A panel of volunteer judges will judge each entry on the following equally weighted criteria: novelty, feasibility of implementation, technical rigor, and how well the concept is presented. Cash prizes for top three entries of US$1000, US$500, and US$300 per team will be awarded.

Session 3aSPb

Signal Processing in Acoustics and Psychological and Physiological Acoustics: Cognitive Signal Processing

Grace A. Clark, Chair
Grace Clark Signal Sciences, 532 Alden Lane, Livermore, CA 94550

Chair's Introduction—9:00

Invited Papers

9:05

3aSPb1. Random linear packet coding for half duplex underwater channels. Rameez Ahmed and Milica Stojanovic (ECE, Northeastern Univ., 360 Huntington Ave., Boston, MA 02115, rarameez@ece.neu.edu)

Random linear packet coding is considered for underwater acoustic communications in situations where reliability is essential, i.e., where data packet delivery has to be guaranteed with some probability. We develop an adaptive power and rate control strategy for a half duplex acoustic channel in which the receiver can inform the transmitter about the channel state. We consider transmission in cycles, each long enough to conform to the channel's coherence. In each cycle, the receiver feeds back the channel gain and the noise power, and the transmitter adjusts its settings accordingly. Based on the experimental measurements, we model the channel gain as a log-normally distributed fading process, and provide a system optimization procedure based on a chosen criterion, such as minimum average energy per bit or maximum average throughput. System performance is compared analytically and numerically to that of a non-adaptive system, showing benefits of closed-loop, channel-aware transmission.

9:25

3aSPb2. Non-Gaussian correlated process sampling for Bayesian cognitive target classification. Grace A. Clark (Grace Clark Signal Sci., 532 Alden Ln., Livermore, CA 94550, clarkga1@comcast.net)

A cognitive signal processing system (for example, in radar or sonar) is one that observes and learns from the environment; then uses a dynamic closed-loop feedback mechanism to adapt the illumination waveform so as to provide system performance improvements over traditional systems. Current cognitive radar algorithms are designed only for target impulse responses that are Gaussian distributed

to achieve mathematical tractability. Our research generalizes the cognitive radar target classifier to deal effectively with arbitrary non-Gaussian distributed target responses. Given exemplars of target impulse responses, our Bayesian illumination waveform design algorithm requires the ability to draw complex correlated samples from a target distribution specified by *both* an arbitrary desired probability density function and a desired power spectral density. This capability is realized using kernel density estimation and an extension of a new simple and efficient nonlinear sampling algorithm by Nichols *et al*. Simulations using non-Gaussian target impulse response waveforms demonstrate very effective target classification performance. We discuss practical issues with the application of the algorithms to real-world problems.

Contributed Papers

9:45

3aSPb3. Determination of material parameters to recreate realistic audio quality for sounding materials in virtual sound reproduction based on modal analysis. Muhammad Imran and Jin Yong Jeon (Architectural Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, mimranh1@gmail.com)

The modal synthesis method has been investigated to synthesize realistic virtual sounds for rigid bodies. The current methods to create realistic audio quality of sounding materials lack the ability to automatically determine material parameters. In this study, an approach is introduced to estimate material parameters that capture the inherent quality of sounding materials and extract perceptually significant features from the recorded example impulse responses. This approach is based on modal analysis using extracted material features to identify the best material parameters for linear modal synthesis (LMS). The synthesized audio produced from LMS preserves the same quality of material sound as recorded sound clips. A perceptual study was also conducted, revealing that the results of this approach are comparable with real recorded sounds in terms of perception of the materials.

10:00

3aSPb4. Decision-feedback equalizer based on channel estimation in the signal subspace. Tsih C. Yang (Dept. of Information Sci. and Electron. Eng., Zhejiang Univ., 38 Zhe Da Rd., Hangzhou, Zhejiang 310027, China, tsihyang@gmail.com) and San Ho Huang (College of Marine Sci., National Sun Yat-sen Univ., Kaohsiung, Taiwan)

The performance of a channel-equalizer is determined by how well the equalizer knows about the channel. For a decision-feedback equalizer based on explicit estimation of the channel impulse response (CIR), the symbol soft-decision-error (SDE) can be directly linked to the channel-estimation error. Due to channel fluctuation and the presence of noise, conventional channel-estimation lacks the required accuracy to achieve minimal SDE (< -10 dB) for a single-receiver channel. One of the approach is to use spatial diversity, where the (effective) CIR for maximum-ratio combining of the multiple-channels, often referred to as the q function, is well-structured and behaved, and relatively accurately estimated. An alternative approach, as proposed in this paper, is to explore the signal subspace for a single-receiver channel, when the multi-paths are cross-correlated as in many underwater acoustic propagation channels. Since the signal occupies a small fraction of the total space, tracking the signal in the subspace has been shown to yield much smaller (by 10–15 dB) signal prediction error (the error between the received-data and predicted-data based on the estimated CIR) and SDE in the training mode. This paper extends the method to the decision mode for a semi-time invariant channel.

10:15

3aSPb5. Improving headphone spatialization for stereo music. Muhammad Haris Usmani (School of Music, Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, usmani@cmu.edu), Ramón Cepeda Jr., Thomas M. Sullivan (Dept. of Elec. and Comput. Eng., Carnegie Mellon Univ., Pittsburgh, PA), and Bhiksha Raj (School of Comput. Sci., Carnegie Mellon Univ., Pittsburgh, PA)

Music is mixed and mastered for playback on near- and far-field speakers, which presents a problem to the growing population of listeners who listen to music primarily on headphones. Playing legacy stereo mixes on headphones places the stereo image inside the listener's head and makes the image appear ultra-wide. While this is helpful for separating the center and stereo content, it has a detrimental effect on the spatialization of the music. It makes headphone listening unnatural and uncomfortable, although listeners have learned to accept it. In this work, we develop a system that processes stereo signals to provide a better sound image to headphone listeners. The sound image is improved by adding the necessary cues to the signal so as to externalize the perceived soundstage by making it similar to the soundstage experienced inside a mixing studio. In order to work with all headphones, our system tries to maintain the mastering equalization curve of the original stereo content. We employ head-related transfer functions, de-correlation, models of reverberation, and 1/3-octave-band equalization to realize our system. The results of pilot subjective evaluations suggest that our system makes music more natural and comfortable to listen to, although with some loss of quality.

10:30

3aSPb6. Multi-modal mapping of complex data using interactive visual and aural cues. JoAnn C. Kuchera-Morin (Media Arts and Technol. (MAT), California Nanosystems Inst., Univ. of California, Santa Barbara, Santa Barbara, CA 93106, jkm@create.ucsb.edu)

At the AlloSphere Research Facility at the University of California, Santa Barbara, our research encompasses multimodal mapping of big and complex data sets. Interactive visual and sonic cues are used to navigate through this complex information. Sonic cues are used to reinforce visual cues as well as used independently, depending on the nature and structure of the data being mapped. In this presentation, I will discuss various research projects and their different or similar mapping configurations depending upon the information being extracted form the data set. We have found that sonic representation is essential for expanding our field-of-view of a large and/or complex data set, as information outside of our visual field-of-view can be sonically represented, thus facilitating visual navigation to that area of the data.

Session 3aUW

Underwater Acoustics: Historical Perspectives on the Origins of Underwater Acoustics III

David L. Bradley, Cochair
Penn State University, PO Box 30, State College, PA 16870

Thomas G. Muir, Cochair
Applied Research Laboratories, University of Texas at Austin, P/O. Box 8029, Austin, TX 78713

Chair's Introduction—9:00

Invited Papers

9:10

3aUW1. Clay and Medwin: An intuitive approach to the math and physics. Mohsen Badiey (Univ. of Delaware, 261 S. College Ave., Newark, DE 19716, badiey@udel.edu) and Tim Stanton (Woods Hole Oceanographic Inst., Woods Hole, MA)

It is interesting to look back to the first encounter with a field that one would eventually make a career. This is frequently a combination of a mentor and some initial learning in an academic environment. Clarence ("Clay") Clay and Herman ("Hank") Medwin provided such an experience for numerous researchers in the field of ocean acoustics. Being active researchers and mentors, they published a sequence of two books, both entitled *Acoustical Oceanography* (1977 and the revised version in 1998), to introduce the field to those interested in using sound to remotely sense the oceans' interior and boundaries. In addition to acousticians, readership includes marine biologists, geologists, and physical oceanographers. Although each book may look relatively simple at first glance, they present an intuitive approach to complex ocean acoustic problems with the right balance between physics, math, and real-world applications. There are also many references to more in-depth treatments of the topics. Each book provides a range of diverse topics including principles of underwater sound, sonar systems, signal processing, nonlinear acoustics, scattering from objects and creatures in the sea, the seafloor, acoustic waveguides, and many other subjects.

9:30

3aUW2. Six decades of evolution in Underwater Acoustics at the Applied Physics Laboratory, University of Washington. Kevin L. Williams, Dajun Tang, Peter H. Dahl, Eric I. Thorsos, Darrell R. Jackson, and Terry E. Ewart (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, williams@apl.washington.edu)

Professor Joe Henderson of the University of Washington physics department formed the Applied Physics Laboratory during WWII. The lab's initial efforts were to redesign the magnetic influence exploders used in US torpedoes. One of the lab's first Underwater Acoustics (UA) successes was development of transducers used in the Bikini Atoll Able test (1946). Those transducers, used to trigger other instrumentation, proved critical. Combining UA and torpedo expertise brought APL-UW to the forefront of tracking range design, construction and deployment in Dabob Bay, Nanoose, and St. Croix in the 1950s and 1960. Understanding the torpedo behavior seen in tracking ranges required measuring both the ocean environment and the acoustics within that environment. Making those measurements, as well as development and testing of models based on those measurements, also became standard operating procedure at APL, led in the 50's by Murphy and Potter. This blueprint of applied research motivating basic research, and the pursuit of basic research via ocean experiments and high fidelity modeling, continues to this day. The presentation will follow this evolution. APL-UW ocean experiments carried out during that time, as well as notable APL-UW research papers, technical reports, computer codes and textbooks, will be used as guideposts.

9:50

3aUW3. Underwater acoustics at Applied Research Laboratories, The University of Texas at Austin. Thomas G. Muir, Clark S. Penrod, and Chester M. McKinney (Appl. Res. Labs., The Univ. of Texas at Austin, P/O. Box 8029, Austin, TX 78713, muir@arlut.utexas.edu)

Several University of Texas physicists and acousticians served at the Harvard Underwater Sound Laboratory during World War II. Among them was Associate Director Paul Boner who founded Applied Research Laboratories (ARL:UT) upon his return to Texas in 1945. The underwater acoustics program began in 1949 under sponsorship of the Naval Ordnance Laboratory in support of passive sea mine sensors. The talk will focus on growth of the program in the early years including basic research, advancements in sonar engineering and signal processing, torpedo, mine and mine countermeasures (for both ocean and riverine environments), work on ship signatures, sediment physics and properties, target strength, submarine and anti-submarine sonar for both active and passive modes, *in-situ* calibration of ship transducers, environmental acoustics, long-range propagation and modeling, and nonlinear acoustics. ARL:UT's participation and support of the university's missions of research, education, and public service will also be discussed.

10:10–10:30 Break

10:30

3aUW4. Penn State's Applied Research Laboratory contributions to underwater acoustics. Edward G. Liszka (Appl. Res. Lab., Penn State Univ., P.O. Box 30, State College, PA 16804, egl4@psu.edu)

The Applied Research Laboratory (ARL) at the Pennsylvania State University has had a rich history of research and development in the field of underwater acoustics dating from its founding in 1945. The author reviews some of the important contributions made over the intervening decades in sub-fields such as sound propagation, scattering, transducers and arrays, signal processing, autonomous systems decision making, structural vibration and sound radiation, and hydro acoustics. The Laboratory was established by the US Navy as a derivative of the Harvard Underwater Sound Laboratory following World War II. Dr. Eric Walker, the first Director, was charged with continuing research and development on acoustically guided torpedoes following his highly successful effort during the war. In 1949, the Garfield Thomas Water Tunnel was constructed to further the understanding of flow-driven underwater acoustic phenomena, such as cavitation and flow induced vibration as examples of early achievements. Due to a growing need for scientists and engineers educated in acoustics, the Navy encouraged Penn State to establish its Graduate Degree Program for Acoustics in 1965. While the Program covers a broad scope of acoustic disciplines, it has a served a significant role in the development of the underwater acoustics field.

10:50

3aUW5. Underwater acoustics at the Johns Hopkins University Applied Physics Lab. Bruce K. Newhall and James W. Jenkins (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723, bruce.newhall@jhuapl.edu)

The Johns Hopkins University Applied Physics Lab (JHU/APL) was founded in 1942. JHU/APL began research in underwater acoustics in 1970 and became known for full scale ocean testing. JHU/APL developed the first technique to accurately measure the shape of a towed array. Towed array testing culminated in 1991–1992 with the deployment of a 5 km aperture measuring signal coherence and beam noise statistics in the Pacific, Atlantic, and Mediterranean. In 1982, JHU/APL began investigations into low frequency (LF) active sonar. Initially, airguns and explosives were employed to measure bottom and surface scattering strengths. Then, tests with a stationary controlled source were conducted from 1986–1989, activating both stationary and towed receiver arrays. In 1989, JHU/APL outfitted the Cory Chouest, adding a two story back deck superstructure. The lower level housed a three aperture LF source array and long towed receiver array, while the upper story berthed 50 scientists and engineers. This ship conducted a series of measurements of scattering strengths of the ocean surface, bottom and volume from 200–1000 Hz. Today JHU/APL employs 5000 staff of which about 150 are scientists and engineers working in underwater acoustics. Studies have expanded from LF beginnings to the full spectrum of acoustic frequencies.

Contributed Papers

11:10

3aUW6. A history of the study of low frequency sound absorption in the sea—An active 50 years and it isn't over yet. David G. Browning (139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com), Vernon P. Simmons (335 Burgundy Rd., Healdsburg, CA), and Peter D. Herstein (24 Mohegan Rd., Charlestown, RI)

In 1965, Thorp published a compilation of low frequency data from SOFAR propagation that indicated absorption below 10,000 Hz was greater than predicted. It was suggested this may be an artifact, but the data were shown to fit a 1 kHz relaxation mechanism. NUWC (formerly NUSC) began an extensive at-sea measurement program, ranging from Hudson Bay (jointly with Canadian DREP) to Lake Tanganyika, while several other laboratories obtained data in other oceans. Yeager and Fisher, using the T-jump method at the Case-Western Laboratory, determined there was a 1 kHz relaxation in seawater associated with boron. The data collected at-sea showed, however, a variation between ocean areas, notably values from the North Pacific were 1/2 those in the North Atlantic. This puzzled supporters and encouraged critics. At Scripps, Simmons and Fisher, using a resonating sphere, showed that the boron relaxation did indeed cause the absorption of low frequency sound. Browning and Mellen discovered the key to the at-sea variation was *p*H, and at NUWC Mellen confirmed this dependence in the laboratory using Simmons' sphere. This direct connection between low frequency absorption and ocean acidification has produced several recent papers on possible implications, and increasing global warming suggests an interesting future.

11:25

3aUW7. A historical perspective: Frank Andrews and acoustics at The Catholic University of America. Joseph F. Vignola (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20064, vignola@cua.edu), Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC), Diego Turo, Shane Guan (Office of Protected Resources, National Marine Fisheries Service, Washington, DC), and John Judge (Mech. Eng., The Catholic Univ. of America, Washington, DC)

The Catholic University of America has a long history in the field of acoustics dating back to the 1930s. Some notable acousticians associated with Catholic University during the early development of the acoustics program there include Father Frances Fox, Karl Herzfeld, Theodore (Ted) Litovitz, among many others. Herzfeld and Litovitz are responsible for an enormous amount of pioneering work in the field of acoustic dissipation. Under the direction Litovitz, Acoustics moved from its original home in the Physics Department to Engineering in the early 1960s. Litovitz in turn recruited Frank Andrews. Andrews took the helm of acoustics at Catholic University and expanded the scope of the program from largely research focused to include a broader emphasis on both graduate teaching and developing the critical connection to practical Navy interests. This presentation will describe the history of acoustic research at Catholic University with a focus on Frank Andrews' accomplishments and contributions to Catholic University and the acoustics community at large.

11:40–12:00 Panel Discussion

Session 3pAA

Architectural Acoustics: Uses, Measurements, and Advances in the Use of Diffusion and Scattering Devices

Ronald Sauro, Chair
NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541

Chair's Introduction—1:15

Invited Papers

1:20

3pAA1. Acoustical optimization of curvilinear shapes used in modern architecture. Peter D'Antonio (RPG Diffusor Systems, Inc., 651C Commerce Dr., Upper Marlboro, MD 20774, pdantonio@rpginc.com) and Richard H. Talaske (Talaske | Sound Thinking, Oak Park, IL)

Classic architecture is characterized by beautiful statuary, coffered ceilings, columns, and relief ornamentation. These beautiful features coincidentally also provided excellent sound diffusion. As architecture evolves into using less ornate surfaces, the acoustic fallout is that modern rooms may not have good sound diffusion. As a result, there is a need for scattering surfaces that compliment contemporary architecture, the way that the afore-mentioned surfaces complemented classic architecture. In order to generate these modern sound diffusing surfaces, a software program called the Shape Optimizer was developed utilizing an accurate prediction method, a diffusion coefficient metric to evaluate performance and an intelligent search engine which can quickly and efficiently navigate through the myriad shape possibilities. One of the authors (Talaske) will also present an acoustician's perspective on how to successfully meet the challenge of marrying the acoustical scattering requirements with the aesthetic goals expressed by the architect and/or interior designer. Several examples will be given by both authors illustrating the use of curvilinear diffusors and the potential of providing diffusing and absorbing surfaces with the same topology.

1:40

3pAA2. The effects of materials on the performance of acoustic diffusers. Richard L. Lenz (RealAcoustix LLC, 2637 N. Washington Blvd., #125, N. Ogden, UT 84414, RL@RealAcoustix.com)

Over the years, many different theories about the effects of material mass, finishes, coverings such as fabrics, porosity, and other physical attributes of diffuser design have been put forth. These theories have included excess absorption and/or reduced diffusion being caused by different woods or other materials being used as well as different types of paints and finishes. This paper is a study on the veracity of those theories and the effects of assorted materials in the construction and implementation of acoustic diffusers. The study will include the use of a specific diffuser design utilizing a periodic design with specific frequency ranges better known as quadratic residue diffusers. Tests will be conducted to look at specific absorption results and the correlation to diffusion efficiency, if any. In addition, different styles of QRD's will be utilized to see if any designs can overcome the various attributes in design used in diffusion. Standard and customized testing procedures will be used to derive the information presented in the paper.

2:00

3pAA3. Particle simulations advance insight into diffusion device design, efficiency, and diffuse field development and propagation. James J. DeGrandis (Acoust. First, 2247 Tomlyn St., Richmond, VA 23230-3334, jim@acousticsfirst.com)

After the development of a demonstration to illustrate acoustic diffusion using Ping-Pong balls (DeGrandis, 2011, "Sound Diffusion Explained," http://acousticsfirst.com/educational-videos-acoustic-sound-diffusion.htm), the idea of virtualizing the concept to gain more insight into diffusion from a particle physics perspective became a goal. Using open-source, Ray tracing modeling, and physics simulation software, the efficiency of different diffusor designs was evaluated. Simulations included individual elements and arrays in varied distribution and varied environments, and their impact on the propagation of sound, as well as the visualization and prediction of the development of an acoustically diffuse field. Integrating a simulated particle emitter to model the excitation, environmental physics to model atmospheric conditions, and 3d models with adjustable surface dampening properties, usable simulations with calculable and predictable properties could be run. New data were collected based on spatial diffusion, isometry, and the speed of which the excitation was broken down from a contiguous wave front into a diffuse particle field. Simulations varied from individual unit efficiency to complex room modeling with multiple emitters and scattering surfaces. This is a collection of the findings from that exercise.

2:20

3pAA4. Diffusion and scattering: A critique of existing standards and data. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

This presentation will look at the existing standards used to measure scattering and diffusion. It will point out scientific problems, unreasonable assumptions, and methodology problems in these standards. Scattering has been assumed to be a separate property of unevenly shaped materials, and therefore, a method was needed to "measure" this property. It will be pointed out that scattering is really

a part of the sound absorption characteristics of materials, i.e., diffraction and can be measured as part of the sound absorption characteristic. Scientific problems and unreasonable assumptions in the standard will also be pointed out. Diffusion can be characterized by the directivity of the energy as well as the magnitude and phase of the energy radiating from the diffuser. The methodologies used to create diffusion can be described as geometric or diffractive. The existing standards do not measure the magnitude or phase of the radiated energy. The results derived from the existing standards approximate the energy losses using scattering and the directivity measured is only used to compare diffusers but cannot be used in acoustic simulation programs.

2:40

3pAA5. A better method or methodology for the measurement of diffusers. Ronald Sauro (NWAA Labs, Inc., 90 Tower Blvd., Elma, WA 98541, audio_ron@msn.com)

This presentation is a follow-on to the presentation “Diffusion and Scattering: A critique of existing standards and data (Goodbye, Scattering).” In this presentation, we will present a method of displaying 3D directivity data for diffusers, pointing out the similarities between the directivity data for diffusers and loudspeakers, and how those similarities can be used to answer the problem of creating usable data for acoustical simulation programs. We will point out that other acoustic and electro-acoustic organizations have solved many of the problems with characterizing the spatial characteristics of sound sources and how diffusers can meet those same criteria. These solutions offer a roadmap for us to be able to characterize diffusers as well. We will also present a way of measuring these diffusers using generally accepted methods, and introduce a method of presenting this data in an easily understood way with a format that can be used in acoustic simulation programs.

WEDNESDAY AFTERNOON, 20 MAY 2015 — RIVERS, 1:30 P.M. TO 2:45 P.M.

Session 3pAB

Animal Bioacoustics: Biosonar

Adrienne M. Copeland, Chair
University of Hawaii at Manoa, P.O. Box 1106, Kailua, HI 96734

Contributed Papers

1:30

3pAB1. Validating side scan sonar as a fish survey tool over artificial reefs in the Gulf of Mexico. Michael Bollinger and Richard Kline (Biological Sci. Dept., Univ. of Texas at Brownsville, 1 W University Blvd., Brownsville, TX 78520, michael.bollinger127@utb.edu)

Artificial reef placements are becoming an important part of fisheries management strategies worldwide due to the loss of natural reefs, and in the Western Gulf of Mexico due to the scarcity of hard structures and vertical relief near shore, which are essential habitat for reef fish. Current visual survey techniques can be crippled by low visibility and unpredictable currents in the Gulf of Mexico, but hydroacoustics can provide a solution to these problems. This study focuses on using ground-truthed side scan sonar technology to determine fish community biomass. Through fish abundance surveys and *in situ* fish sampling, we developed a fish survey protocol using side scan sonar to quantify fish assemblages over artificial reefs. The effectiveness of this technology for management purposes was also demonstrated by comparing it with visual census methods.

1:45

3pAB2. A design for a biomimetic dynamic sonar head. Philip Caspers, Yanqing Fu, and Rolf Mueller (Mech. Eng., Virginia Tech, 1110 Washington St., SW, MC 0917, Blacksburg, VA 24061, pcaspers@vt.edu)

The biosonar system of horseshoe bats (family Rhinolophidae) has been shown to employ unusual dynamics upon the emission as well as the reception of the ultrasonic pulses. Non-rigid changes to the shapes of the noseleaves (emission baffles) as well as the outer ears (pinnae, reception baffles) have been demonstrated to affect the properties of the emitted and received ultrasonic signals. These effects have been found in the results of numerical simulations as well as experimentation with physical prototypes. In the present work, a next-generation prototype of a biomimetic sonar head inspired by horseshoe bats is being developed. The goals for this system are to create more comprehensive and life-like dynamic baffle shape geometries as well as a better acoustic coupling between the ultrasonic transducers and the time-variant baffle shapes. Particular attention has been paid to geometry of the transition between nostrils and the noseleaf baffle. A single biomimetic system that incorporates these dynamic emission and reception baffles will enable an experimental investigation of how these two dynamic stages could be used in an integrated fashion to enhance sonar performance in real-world sonar sensing scenarios.

2:00

3pAB3. Echolocation beam shape of the Risso’s dolphin (*Grampus griseus*). Adam B. Smith (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, 1933 Iwi Way, Honolulu, HI 96816, adambsmi@hawaii.edu), Laura Kloepper (Neurosci., Brown Univ., Providence, RI), Wei-Cheng Yang (Veterinary Medicine, National Chiayi Univ., Chiayi, Taiwan), Wan-Hsiu Huang, I-Fan Jen (Farglory Ocean Park, Hualien, Taiwan), and Paul E. Nachtigall (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, Kaneohe, HI)

Studies have shown that echolocation signals of some odontocete species are projected in both single and sometimes vertically dual-lobed beam shapes. In this study, the echolocation beam of a Risso’s dolphin (*Grampus griseus*) was measured from a captive individual ensonifying an underwater target. Clicks were recorded with an array of 16 hydrophones and the two dimensional beam shape was explored using frequency-dependent

amplitude plots. Click source parameters were comparable to those already described for this species, while analysis revealed primarily single-lobed, and occasionally vertically dual-lobed, beam shapes. Center frequencies of click signals increased with increasing sound pressure level, while the −3 dB beam radius decreased with increasing center frequency. This study is the first to measure the beam shape of echolocation signals in *G. griseus*, which exhibits forms similar to those found in the bottlenose dolphin and false killer whale

2:15

3pAB4. The effect of mouth gape angle on beam size for echolocating *Eptesicus fuscus*. Laura Kloepper, Rebecca Wojciechowicz, and James Simmons (Dept. of Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, laurakloepper@gmail.com)

The sonar beamwidth of mouth-emitting echolocating bats is hypothesized to change according to mouth gape angle, with a narrowing of the beam predicted with wider mouth openings. To investigate the relationship between mouth gape angle and beam size, we recorded both the emitted sonar beam as well as the mouth opening for four bats during a target detection task. Sonar signals were recorded with an array of microphones, and bat mouth dynamics was recorded with an action camera recording at 240 frames per second. The video frames corresponding to each pulse emission were extracted and the mouth gape angle was measured. Sound intensity, −3 dB beam angle, and frequency characteristics were calculated from the microphone array and compared to the mouth gape angles to determine the influence of mouth opening on echolocation signals.

2:30

3pAB5. Comparing the biosonar signals of free swimming dolphins with those of a stationary dolphin in a net pen. Whitlow W. Au, Adrienne Copeland (Hawaii Inst. of Marine Biology, Univ. of Hawaii, 46-007 Lilipuna Rd., Kaneohe, HI 96744, wau@hawaii.edu), Stephen W. Martin, and Patrick W. Moore (National Marine Mammal Foundation, San Diego, CA)

The biosonar signals of two free-swimming Atlantic bottlenose dolphins performing a complex sonar search for a bottom target in San Diego Bay were compared with the biosonar signals of a dolphin performing a target discrimination task in a net pen. A bite-plate device that the dolphins carried supported a hydrophone that extended directly in front of the dolphin. A biosonar measuring tootbox (BMT) attached to the bite plate measured the outgoing biosonar signals while the dolphins conducted sonar searches. Both of the free-swimming dolphins used different biosonar search strategy in solving the problem and their biosonar signals reflect the difference in strategy. The dolphin stationed in a hoop in the pen while echolocating on a target 6 m away and reported if the indentation on a spherical target was directed toward it. The signals were parameterized by determining the peak-to-peak source levels, source energy flux density, peaked frequency, center frequency, rms bandwidth, rms duration, and the Q of the signals. Some of the characteristics of the r signals were similar for the free-swimming and stationary dolphins while some were significantly different suggesting biosonar signals used by free-swimming animals may be different than signals used by captive dolphins in restrictive environments.

WEDNESDAY AFTERNOON, 20 MAY 2015 KINGS 1, 1:00 P.M. TO 2:05 P.M.

Session 3pAO

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

Andone C. Lavery, Chair
Applied Ocean Physics and Engineering, Woods Hole Oceanographic Institution, 98 Water Street, MS 11, Bigelow 211, Woods Hole, MA 02536

3p WED. PM

Chair's Introduction—1:00

Invited Paper

1:05

3pAO1. Monitoring deep ocean temperatures using low-frequency ambient noise. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

In order to precisely quantify ocean's heat capacity and ocean's influence on climate change, it is important to accurately monitor ocean temperature variations, especially in the deep ocean (i.e., at depths ~1000 m), which cannot be easily surveyed by satellite measurements. Indeed, to date, deep ocean temperatures are most commonly measured using free-drifting oceanographic floats (e.g., ARGO floats), but this approach only provides a limited spatial and temporal resolution because of the sparseness of the existing global network of oceanographic floats. On the other hand, acoustic thermometry has already been demonstrated as an important modality for measuring ocean temperature and its heat capacity using low-frequency sound. However, current implementations of acoustic thermometry requires the use of active sources; aside from the technology issues of deploying such sources, there is also the ongoing issue of the potential disturbance of marine animals. We will demonstrate a totally passive acoustic method of acoustic thermometry based only on coherent processing of low-frequency ocean noise ($f<50$Hz) and whose results are in agreement with classical point measurements obtained from oceanographic floats. We will discuss how passive acoustic thermometry could improve global monitoring of deep ocean temperature variations through implementation using a global network of hydrophone arrays.

Session 3pBA

Biomedical Acoustics: Biomedical Acoustics Best Student Paper Competition (Poster Session)

Kevin J. Haworth, Chair
University of Cincinnati, 231 Albert Sabin Way, CVC3940, Cincinnati, OH 45209

The ASA Technical Committee on Biomedical Acoustics offers a Best Student Paper Award to eligible students who are presenting at the meeting. Each student must defend a poster of her or his work during the student poster session. This defense will be evaluated by a group of judges from the Technical Committee on Biomedical Acoustics. Additionally, each student will give an oral presentation in a regular/special session. Up to three awards will be presented to the students with USD $500 for first prize, USD $300 for second prize, and USD $200 for third prize. The award winners will be announced at the meeting of the Biomedical Acoustics Technical Committee. Below is a list of students competing, with abstract numbers titles. Full abstracts can be found in the oral sessions associated with the abstract numbers.

All entries will be on display and all authors will be at their posters from 1:00 p.m. to 3:00 p.m.

2aBA3. Radial excursions of bound and non-bound targeted lipid-coated single microbubbles
Student author: Tom van Rooij

2aBA4. Heat and mass transfer effects on forced radial oscillations in soft tissue
Student author: Carlos Barajas

2aBA10. Effect of blood viscosity and thrombus composition on sonoreperfusion efficacy
Student author: John Black

2aBA12. Stress and strain fields produced by violent bubble collapse
Student author: Lauren Mancia

2pBA13. Broadband ultrasound scanning and frequency sweep measurement for trabecular bone with novel wideband single crystal transducer
Student author: Jian Jiao

2pBA14. Ultrasonic wave velocities in radial direction of bovine cortical bone
Student author: Yuma Nishimura

2pBA15. FDTD simulations of ultrasonic wave propagation in the cortical bone with heterogeneity
Student author: Toshiho Hata

3aBA8. Enhancement of osteogenesis and stem cell differentiation by injectable nanocomposite and dynamic acoustic radiation force stimulation
Student author: Vaishnavi Shrivastava

3aBA12. Sternal vibrations reflect hemodynamic changes during immersion: Underwater ballistocardiography
Student author: Andrew D. Wiens

4aBA6. Compensating for Scholte waves in single track location shearwave elasticity imaging
Student author: Jonathan Langdon

4aBA9. Three-dimensional finite difference models of shear wave propagation in isotropic, homogeneous soft tissue
Student author: Yiqun Yang

4pBAa4. Laser-induced vaporization and acoustic-induced cavitation of droplets
Student author: Jaesok Yu

4pBAa5. Numerical investigation of the nonlinear attenuation and dissipation of acoustic waves in a medium containing ultrasound contrast agents
Student author: Amin Jafari Sojahrood

4pBAb2. A sum-of-harmonics time-domain method to distinguish harmonic and broadband signals during passive acoustic mapping of ultrasound therapies
Student author: Erasmia Lyka

4pBAb8. Designing ultrasound fields to control the morphology of engineered microvessel networks
Student author: Eric S. Comeau

4pBAb9. A nonlinear derating method for estimating high-intensity focal pressures in tissue
Student author: Seyed Ahmad Reza Dibaji

5aBA1. Numerical evaluation of absorbing boundary layers for the transient Khokhlov-Zabolotskaya-Kuznetsov Equation
Student author: Xiaofeng Zhao

5aBA2. An improved interpolation approach for rapid simulations of pulse-echo ultrasound imaging
Student author: Leslie P. Thomas

5aBA3. Toward monodisperse ultrasound-triggered phase-shift emulsions using differential centrifugation
Student author: Kyle Stewart

5aBA4. Characterizing the pressure field in a modified microbubble flow cytometer: Using a laser Doppler vibrometer to validate the internal pressure
Student author: Cheng-Hui Wang

5aBA7. On the use of local speckle field as a correction factor for shear modulus estimates based on multiple-track-locations methods
Student author: Laurentius O. Osapoetra

5aBA8. Iterative reconstruction of the ultrasound attenuation coefficient from backscattered signals
Student author: Natalia Ilyina

5aBA11. Three-dimensional pulsation of rat carotid artery bifurcation observed using a high-resolution ultrasound imaging system
Student author: Changzhu Jin

5aBA13. Novel use of ultrasound imaging to decode activity of forearm muscles for upper extremity prosthetic control
Student author: Nima Akhlaghi

WEDNESDAY AFTERNOON, 20 MAY 2015 BALLROOM 4, 1:00 P.M. TO 2:05 P.M.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Michelle C. Vigeant, Chair
Graduate Program in Acoustics, The Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

3p WED. PM

Chair's Introduction—1:00

Invited Papers

1:05

3pID1. Hot topics in speech communication: Acoustics of regionally accented speech. Robert A. Fox and Ewa Jacewicz (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu)

Pronunciation patterns of native speakers of American English vary across the country in a systematic way, being shaped in part by social and cultural factors in a particular geographic area. The detectable regional (or dialect) variation in speech can considerably alter speech recognition and perceptual processing. The intelligibility of regionally accented speech has its source in acoustic variation present in dynamic sound structures, which convey an unprecedented richness of acoustic phonetic details along multiple dimensions in space and time. The challenge for speech communication research is to understand which complex acoustic structures are essential in shaping auditory sensitivity to regional variation and which are coexistent and necessarily redundant. New directions in this research area include acoustic explorations of complex interactions among spectral, source, and temporal components whose unique balance is modified by regional and social influences. Recent advances in these acoustic explorations will be presented.

1:25

3pID2. Hot topics in noise. Scott D. Sommerfeldt (Dept. of Phys., Brigham Young Univ., N181 ESC, Provo, UT 84602, scott_sommerfeldt@byu.edu)

There are a number of diverse areas of interest in TC Noise. Many of these have developed in response to adverse human reaction to various noise sources. This paper will briefly overview some of these current topics to give some exposure to what is happening in "Noise." These include wind turbine noise, which has become of concern with the development of wind farms near residents; soundscapes, which is focused on engineering a desirable environment in the settings where we live; high amplitude jet noise, which is becoming of greater concern as source levels increase, and humans and animals are exposed to higher sound pressure levels; and active noise control, which continues to mature and is beginning to show up in more applications in society.

1:45

3pID3. Hot topics in physical acoustics. Joel Mobley (Phys. and Astronomy/NCPA, Univ. of MS, PO Box 1848, 1034 NCPA, University, MS 38677, jmobley@olemiss.edu)

From the astronomical to the molecular, acoustics is central to phenomena across a vast spectrum of length scales. This talk provides an overview of recent developments in physical acoustics across the size continuum, while looking more in depth in three areas. At the larger end of the scale, the generation of the infrasound by cyclonic storms is considered, specifically looking at the infrasonic signatures that can be used to track and characterize these systems. At intermediate length scales, advances in acoustic metamaterial (AM) research are examined. Specific examples will include energy-harvesting metasurfaces and soft matter systems. Moving to the small end of the scale, phoxonic crystals (PxC) are discussed. PxC's possess both optical and acoustic band gaps and can be used to trap light and sound together, enhancing the strength of acousto-optic interactions.

WEDNESDAY AFTERNOON, 20 MAY 2015 — KINGS 2, 1:00 P.M. TO 2:20 P.M.

Session 3pMU

Musical Acoustics: General Topics in Musical Acoustics I

Randy Worland, Cochair
Physics, University of Puget Sound, 1500 N. Warner, Tacoma, WA 98416

Whitney L. Coyle, Cochair
The Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

Jack Dostal, Cochair
Physics, Wake Forest University, P.O. Box 7507, Winston-Salem, NC 27109

Chair's Introduction—1:00

Contributed Papers

1:05

3pMU1. Acoustic characteristics of a guitar. Corey A. Taylor (Graduate Program in Acoust., The Penn State Univ., 130 Famstead Ln., Apt. #171, State College, PA 16803, cat240@psu.edu) and Daniel A. Russell (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

A variety of tests were done on an acoustic guitar to help characterize and quantify the radiated sound. Modal analysis was performed to determine resonant low frequency modes of the guitar and the motion of the air in the sound hole. Mechanical impedance and mobility measurements were taken at the bridge. String decay and radiated decay times were measured at several fret positions to study the interaction between modal resonances and decay times. Many of these individual tests have been performed on acoustic instruments before. This paper will combine the analysis of all tests and measurements to show that the modal frequencies play a large role in affecting the sound quality of an acoustic guitar.

1:20

3pMU2. Using spectral analysis to evaluate flute tone quality. Ron Yorita (Comput. Sci., California Polytechnic State Univ., PO Box 834, Morro Bay, CA 93443, ron.yorita@gmail.com) and John Clements (Comput. Sci., California Polytechnic State Univ., San Luis Obispo, CA)

Many skilled flutists place a high priority on "good" tone quality, or timbre. Unlike pitch and rhythm, timbre is difficult to objectively quantify. This project explores (1) how tone quality is described by skilled flutists, (2) whether the harmonic spectrum has some correlation with tone quality, (3) whether certain harmonic spectra are preferred, or considered "good." Thirty-one flutists ranging from high school students to professionals were recorded. A set of samples was used in surveys and interviews to capture descriptors and ratings of tone quality. All of the recorded samples were analyzed using application programs, Harmonic Analysis Tools (HAT), created for this study. HAT uses digital signal processing techniques to produce "spectral signatures". The signatures consist of the harmonic

content, pitch, and amplitude of a sample. The outcome of this research is a baseline set of some often used descriptors. In addition, results showed some correlation between harmonic spectra and descriptors. There were also trends in preferences with respect to certain spectral characteristics. An unexpected finding was that University students showed divergent timbre preferences compared to highly experienced flutists.

1:35

3pMU3. A full double tuba in F/BBb. Frederick J. Young (Communications and Arts, Univ. of Pittsburgh at Bradford, 800 Minard Run Rd., Bradford, PA 16701, youngfj@youngbros.com)

Described here is a design for an F/BB^b tuba. The valves and keys needed for a full double tuba take a lot of space. There exists a full and complete double tuba in BB^b/EEE built on a very large instrument frame and is more than 5 feet tall and 2 feet wide [1]. However, the sizes of F tuba frames are too small to accommodate 5 double and 2 switch valves and their associated tubing. Using a standard F tuba frame, the lack of space has been solved by having four double valves and two switch valves. The fourth double valve is unusual because it is descending on the F side and ascending on the BB^b side of the instrument. An ascent to a BB □tuba is made by the ascending part of that valve. The valve slide lengths have been optimized to yield intonation such that $-5<$ intonation error < 5 cents over a five semitone descent from an open note. Various acoustic measurements are reported for the instrument which will be exhibited. [1] F. J. Young, "Five valve compensating brass wind instrument," Proc. Meetings Acoust. **11**, 035004 (2011).

1:50

3pMU4. Experimental investigation of individual musicians on tonal quality for saxophones and clarinets. Charles E. Kinzer (Dept. of Music, Longwood Univ., Farmville, VA 23901, kinzerce@longwood.edu), Walter C. McDermott, and Stanely A. Cheyne (Phys. and Astronomy, Hampden-Sydney College, Hampden-Sydney, VA)

An experimental investigation designed to quantify the effects of individuals on the tonal quality of a given saxophone, mouthpiece and reed combination, as well as a clarinet, mouthpiece, and reed combination has been performed. Several musicians, using the same instrument of each configuration, played a single note from each of these instruments which was digitally recorded. A Fourier transform of the recorded waveforms was used to compare the frequency spectrum from each musician. The results of the comparison will be reported and discussed.

2:05

3pMU5. Air columns for demonstration of upstream/downstream symmetry in reed wind instruments. Peter L. Hoekje (Phys. and Astronomy, Baldwin Wallace Univ., 275 Eastland Rd., Berea, OH 44017, phoekje@bw.edu)

The acoustics of a reed wind instrument such as a clarinet or oboe, and including the lip-reed instruments such as the trumpet, are classically described by the input impedance of the instrument air column and the dynamical flow control characteristics of the reed valve. The behavior of the system is largely dependent on the resonances of the instrument air column. A fuller explanation however includes both the downstream (usually the instrument) and the upstream (usually the player's wind way) input impedances in symmetry. To demonstrate this, a clarinet-like instrument with no resonance peaks can be played using the resonances of the player's wind way. Another demonstration uses a resonant tube for the upstream air column, with the playing frequency or fundamental based upon a resonance of this upstream tube. The downstream air column has a resonance whose frequency and damping can be set independently of that other resonance. This allows exploration of the effect of a resonance at a harmonic of the playing frequency.

WEDNESDAY AFTERNOON, 20 MAY 2015 KINGS 5, 2:00 P.M. TO 3:15 P.M.

Session 3pNS

Noise: Environmental Noise and Noise Control Elements

Peter Newman, Chair
Recreation, Park and Tourism Management, The Pennsylvania State University, University Park, PA 16802

Contributed Papers

2:00

3pNS1. A spatially explicit estimate of environmental noise exposure in the contiguous United States. Daniel Mennitt (Elec. and Comput. Eng., Colorado State Univ., 1201 Oakridge Dr., Ste. 100, Fort Collins, CO 80525, daniel_mennitt@partner.nps.gov), Kurt Fristrup, and Lisa Nelson (National Park Service, Fort Collins, CO)

Environmental noise is widespread across the United States, the spatial patterns of which are dependent on a complex linkage of environmental and socioeconomic factors. Chronic exposure brings with it adverse consequences to terrestrial organisms; effects on human health and well-being include hypertension, cardiovascular disease, sleep disturbance, cognitive impairment, and annoyance. Assessments of noise exposure are essential to understand the extent of impact as well as inform land use planning and noise abatement strategies. Using extensive empirical data and a geospatial framework, we modeled the day-night average sound level (L_{dn}). The dominant factors driving sound levels are land use, climate, population, and proximity to traffic corridors. Model predictions were mapped to reveal the spatial distribution of expected sound pressure levels across the contiguous United States at a resolution of 270 m. The expected L_{dn} was compared with localized population density to estimate the number of inhabitants exposed to levels that put them at increased risk for non-auditory health effects. The magnitude and extent of noise exposure suggests a substantial opportunity to enact measures that will improve the quality of life for many Americans.

2:15

3pNS2. Visitor assessment of anthropogenic noise at Grand Canyon National Park. Pranav K. Pamidighantam and Paul Schomer (Schomer and Assoc., Inc., 2117 Robert Dr., Champaign, IL 61821, ppamidig@illinois.edu)

In the last decade, multiple studies have been conducted at national parks using the traditional uni-polar annoyance scale and the dose-response concept to determine the effects of aircraft noise on park visitors' experiences. This survey instead uses a bipolar "pleasantness" scale and breaks each visitor's hike into segments so as to understand the efficacy of the pleasantness scale as a metric for visitor perceptions of aircraft noise. This survey was conducted on three different trails at the Grand Canyon: the Hermit trail, a backcountry trail with a large amount of aircraft noise; the Widforss trail, a backcountry trail with minimal aircraft noise; and the Bright Angel trail, a maintained trail with a small amount of aircraft noise but many hikers. Many visitors began hiking before aircraft started flying so the use of segments allowed for a longitudinal study between segments with and without aircraft noise. Through the use of these segments, correlations are drawn and regression models fit between pleasantness and various metrics, such as Leq or percent time audible. The results to date show "pleasantness" to be a viable alternative metric to quantify visitor perceptions of aircraft noise in national parks.

2:30

3pNS3. The impact of anthropogenic noise on wetland habitats. Adrienne Hopson-Costa, Regis Antonioli, Julia Crispim da Fontoura, and Ferenc de Szalay (Dept. of Biological Sci., Kent State Univ., 256 Cunningham Hall, Kent, OH 44242, ahopson2@kent.edu)

Impacts of anthropogenic noise is an important but poorly studied aspect of wetland ecology. Many aquatic animals use sound to attract mates, coordinate movements, defend territory, and detect predators. However, anthropogenic noise (e.g., from boat or automobile motors) are increasingly important in wetlands. Therefore, it is important to understand how anthropogenic noise pollution affects these habitats. I sampled wetland soundscapes in urbanized and rural areas in northeastern Ohio, and used Raven software to analyze the data. Above the water line, average sound power was 38.9db–73.9db, which was similar to below the water line (48.6db–78.5db). Frequencies in the band 0–5 kHz had the highest overall power both above and below the water. One of the most urbanized sites had above/below readings as high as 83.9/79.3 dB. In comparison, one of the most rural sites had readings as low as 47.6/66.8 dB. At both urbanized and rural sites, the 0–5 kHz band had the highest overall power. I further tested if wetland invertebrates could be impacted by anthropogenic sounds. I found that *Procambarus acutus* crayfish produced audible clicks (frequency range 6 kHz–45 kHz, average power 69–73.5 dB). Ongoing laboratory research is testing if motorboat noise affects *P. acutus* behavior and sound production.

2:45

3pNS4. Bayesian inverse analysis of multilayer acoustic porous media. Cameron J. Fackler (3M Personal Safety Div., 7911 Zionsville Rd., Indianapolis, IN 46268, cameron.fackler@mmm.com), Kirill V. Horoshenkov (Dept. of Mech. Eng., Univ. of Sheffield, Sheffield, United Kingdom), and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Porous materials provide sound absorption and noise control in various applications. In many scenarios, different porous materials may be combined into multilayer absorbers to enhance the absorptive properties. We apply the Bayesian inference framework to analyze such multilayer porous materials, developing a method to determine simultaneously the number of constituent layers as well as the physical properties of each layer in a multilayer porous material. The model-based analysis combines a measurement of the acoustic surface impedance or absorption coefficient of a potentially multilayered material sample with a transfer-matrix formulation of multilayer porous material acoustic propagation models. For each sample to be analyzed, the number of layers considered in the propagation model is varied, and Bayesian evidence is computed for each case. Selecting the model with the highest evidence parsimoniously determines the number of layers present in the sample. Once the number of layers has been determined, Bayesian parameter estimation inversely determines the physical properties of each layer by estimating the input parameters of the multilayer propagation model. The proposed method automatically determines the number of layers and physical parameters of a multilayer material without any a priori knowledge of these values.

3:00

3pNS5. Modeling of acoustic resonators and resonator arrays for use in passive noise control. Matthew F. Calton and Scott D. Sommerfeldt (Brigham Young Univ., 266 N. 300 E. #26, Provo, UT 84606, mattcalton@gmail.com)

Acoustic resonators, such as the Helmholtz and quarter-wave resonator, can be used to attenuate unwanted noise in a space. Classic formulations can be used to approximate resonator performance for a given resonator configuration, but may lack sufficient accuracy for some applications. This research aims to improve the analytical characterization of resonators to provide better correlation to experimental results. Using higher order approximations and proper end corrections, more accuracy can be obtained in calculating the impedance and resonance frequency of a single resonator, which will then carry over into the overall configuration of the model. The impedance of a system of resonators in parallel is also considered, where the effects of acoustic coupling can be observed. Resonators with complex, non-ideal geometries are explored for applications where space is limited.

Session 3pPP

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

Michael G. Heinz, Chair
Speech, Language, and Hearing Sciences & Biomedical Engineering, Purdue University, 500 Oval Drive, West Lafayette, IN 47907

Chair's Introduction—1:15

Invited Paper

1:20

3pPP1. Relating physiology to perception: The case of the notched-noise masker. Laurel H. Carney (Biomedical Eng., Univ. of Rochester, 601 Elmwood Ave., Box 603, Rochester, NY 14642, Laurel.Carney@Rochester.edu)

Sharpness of peripheral tuning is an essential factor in auditory signal processing. The notched-noise (NN) masking paradigm (Patterson, 1976, JASA) is an ingenious strategy used in numerous studies to estimate peripheral tuning bandwidths, filter shapes, and level-dependence in normal and impaired ears. Interpretation of NN results is based on the power-spectrum model of masking; however, the fact that NN thresholds are little influenced by a roving-level paradigm calls this interpretation into question (Lentz *et al*., 1999, JASA). Here, we explore neural cues for detection in NN in auditory-nerve (AN) and midbrain (inferior colliculus, IC) responses. Strong low-frequency fluctuations in AN discharge patterns associated with sharp spectral edges provide effective inputs for midbrain neurons tuned to low-frequency fluctuations. Addition of a tone target reduces the fluctuations in some frequency channels, creating a strong contrast in the fluctuation profile along the frequency axis. We explore the NN paradigm using computational models for AN and amplitude-modulation (AM) tuned IC neurons. Recordings from band-pass AM tuned neurons in the IC of awake rabbit support the hypothesis that the midbrain response profile can explain perception in the NN paradigm. Interpretation of NN masking results should include not only peripheral tuning but also central processing mechanisms.

Session 3pSA

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

Kuangcheng Wu, Cochair
Ship Survivability, Newport News Shipbuild, 4101 Washington Ave., Newport News, VA 23693

Jeffrey Cipolla, Cochair
Weidlinger Associates, Inc., 1825 K St. NW, Suite 350, Washington, DC 20006

Contributed Papers

1:00

3pSA1. Vibration analysis of damped stiffened beams and plates by complex modes and frequencies. Giora -. Rosenhouse (swantech, 89 Hagalil Str., Haifa 3268412, Israel, fwamtech@bezeqint.net)

Stiffened beams and plates are used in design of structures forced by machinery excitation, which result in flexural motion having distorted normal modes due to the effect of both stiffeners and damping. Such plates and beams are often damped externally or internally, which leads in turn to complex modes and natural frequencies. The modes, which are normal to the direction or axis of the stiffeners can be calculated by different techniques, such as the "three moments" method for stiffened plates or beams. The effect of the stiffeners is given by finding the changes in frequencies and normal modes as related to the ratio of the stiffeners'- and plate's or beam's rigidities. The paper gives insight into complex modes and the eigenvalue for each such mode and also their relation to the common definition of modal shapes magnitude and frequencies and concludes with solved examples for stiffened plates and beams models.

1:15

3pSA2. Acoustical field and vibration in hearing aid devices. Michael Salameh (Starkey Hearing Technologies, 6600 Washington Ave. S., Eden Prairie, MN 55344, michael_salameh@starkey.com)

One of the main considerations in the hearing aid design is the feedback from the receiver (speaker) to the microphone. Transducer characteristics, mechanical design, material properties, and test setup are among the factors that affect the feedback in hearing aid devices. Characterizing the vibration and the acoustical field generated from different components of the hearing aid device is one of the main challenges in the feedback study and control. In this paper, the acoustical field and the vibration in a hearing aid device are evaluated. Simulation and measurement data are presented and analyzed.

1:30

3pSA3. Resonant ultrasound spectroscopy with large attenuation and arbitrary geometry. Marcel Remillieux, TJ Ulrich, and Pierre-Yves Le Bas (Los Alamos National Lab., Geophys. Group (EES-17), Mail Stop: D446, Los Alamos, NM 87545, mcr1@lanl.gov)

Resonant ultrasound spectroscopy (RUS) is a powerful and established technique for measuring material properties. The first step of this technique consists of extracting resonance frequencies and attenuations from the vibrational frequency spectrum measured on a sample with free boundary conditions. An inversion technique is then used to retrieve the elastic tensor from the measured resonance frequencies. As originally developed, RUS has been mostly applicable to (1) weakly attenuating media where each resonance frequency can be clearly identified and (2) relatively simple geometries where analytical solutions exist. In this presentation, the possibility of using RUS in a highly attenuating medium and on a sample of arbitrary geometry is explored. The Kumaresan-Tufts algorithm is used to fit a sum of exponentially damped sinusoids with closely spaced frequencies to the measured frequency spectrum. The inversion of the elastic tensor is achieved with a genetic algorithm, which allows searching for a global minimum within a discrete and relatively wide solution space. The accuracy of the proposed approach is evaluated against numerical data for which the solutions are known *a priori*. [This work was supported by the U.S. Dept. of Energy, Fuel Cycle R&D, Used Fuel Disposition (Storage) campaign.]

1:45

3pSA4. High order modal approximation techniques in the frequency domain. Andrew S. Wixom and James G. McDaniel (Mech. Eng. Dept., Boston Univ., Boston, MA 02215, awixom@bu.edu)

This paper presents high order modal methods that increase the accuracy of the standard modal truncation scheme with an emphasis on frequency-domain accuracy. The existing Mode Acceleration and Modal Truncation Augmentation methods are compared alongside a new interpretation utilizing the interpolating polynomial. This technique is motivated by taking limits of the modal sum, introducing a residual that appears as a Laurent polynomial. The results presented focus on the case when the frequency range of interest is in the middle of the system's spectrum of eigenvalues, meaning that at least some of the system's natural frequencies lie below the lower bound of the frequency range. This case allows for a possible error in the Modal Truncation Augmentation method that is demonstrated here. Also, convergence properties of the methods are discussed and demonstrated.

2:00

3pSA5. "Negative" acoustic signal propagation in a media with resonant density and compressibility response functions. Valentin Burov, Konstantin Dmitriev, and Sergei Sergeev (Phys., Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, sergeev@aesc.msu.ru)

The properties of acoustical double-negative media strongly depend on the frequency of a propagating wave. Particularly, the effective density and compressibility of a medium can be negative only at the certain frequency band. So it is very important to consider nonstationary processes in such media. The numerical simulations of the medium characterized by resonant-type density and compressibility response functions was performed in the article. It was based on the deduced matrix equation of a Lippmann-Shwinger type. The layer of "resonant" medium was placed inside an infinite background with constant density and compressibility. The incident impulse had a gauss form modulated by a certain carrier frequency. The effective density and compressibility of the layer change from both positive to both negative values while the impulse propagates through it. The phase

velocity also becomes negative and the group velocity remains positive satisfying the causality principle. In the case of long enough impulse duration the layer behaves like it is made of a double-negative medium.

2:15

3pSA6. Vibration isolation design of railroad tracks within Ankara high speed train station. Salih Alan and Mehmet Caliskan (Dept. of Mech. Eng., Middle East Tech. Univ., Ankara 06800, Turkey, caliskan@metu.edu.tr)

Ankara serves as a hub for high speed railway operations within Turkey. A new station building of nine stories in total, which also incorporates a shopping mall and a hotel inside, is currently under construction. Need arises for controlling vibrations due to passing trains through the building. Finite element model of the building is developed by commercial software. The interaction of the building with the foundation is represented by a number of spring and damper combinations whose characteristics are obtained from the existing knowledge of soil-structure interactions. The dynamic stiffness characteristics of isolating layer under the railway slabs are sought. The FRF's between several floors of the building and the base of the track are obtained from the finite element model, to be coupled with FRF's of the train-track-isolating layer system. Vibration isolation design is carried with respect to vibration criteria in Turkish environmental noise regulations.

2:30

3pSA7. Reduction of low frequency scattering from a cylindrical elastic structure using a multi-element multi-path design. David Raudales and Donald B. Bliss (Mech. Eng., Duke Univ., Edmund T. Pratt Jr. School of Eng. Duke University, Box 90300, Hudson Hall, Durham, NC 27708, dr78@duke.edu)

Acoustic scattering of an incident plane wave off an elastic cylinder can be reduced by a multi-element multi-path (MEMP) design. Such a design replaces a single structural element with a system of elastically coupled subsystems, and utilizes the inherent mechanical properties of these coupled substructures, rather than damping, to tune its dynamic response. MEMP structures provide an innovative approach by increasing the number of wave transmission paths in this higher order system, thereby introducing physical phenomena not present in single element structures. Previous analytical and experimental work on vibration transmission in MEMP thin beams and shells demonstrates substantial wide-band attenuation. Current research considers the application of the method to underwater scattering from 2-D concentric thin cylindrical shells with azimuthally continuous radial elastic coupling. Modal decomposition of the thin shell equations is employed for mode-by-mode impedance matching between the Bessel function harmonics of the incident fluid planar wave and the polynomial functions of the mechanical structure. Quasi-static spring and mass impedance matching with the monopole and dipole modes ensures scattering reduction at lower frequencies, while novel resonance matching between the fluid and structure extends reduction into higher frequency ranges. Very promising results reveal possible applications in underwater naval submersibles.

2:45

3pSA8. Refinements to the relaxation model for sound propagation in porous media. D. K. Wilson (U.S. Army Cold Regions Res. and Eng. Lab., Engineer Res. and Development Ctr., 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil)

The relaxation formulation for sound propagation in porous media [D. K. Wilson, *J. Acoust. Soc. Am.* **94**(2), 1136–1145 (1993)] was intended to provide a phenomenologically correct model with relatively simple equations. This paper presents several refinements, which connect the relaxation model more clearly to others in the literature, and clarify interpretation of the parameters. In particular, the relaxation function is applied to the dynamic tortuosity (rather than the inverse of the complex density), and the vorticity and thermal relaxation times are related directly to the length scales of Allard and Champoux [J.-F. Allard and Y. Champoux, *J. Acoust. Soc. Am.* **91**(6), 3346–3353 (1992)]. When formulated in this manner, the relaxation and Allard-Champoux models coincide exactly in the low-frequency/small-pore and high-frequency/large-pore limits, although interpolation between these limits differs somewhat. The revised relaxation model can be readily transformed into a causal time-domain model, involving convolutions between the acoustic fields and a relaxation response function. The basic formulation has five parameters: porosity, static flow resistivity, tortuosity, and two relaxation times. A sequence of model reductions is described, leading finally to two-parameter (porosity and static flow resistivity) equations for the impedance and complex wavenumber, which provide a useful generalization of the Delany-Bazley equations.

WEDNESDAY AFTERNOON, 20 MAY 2015 COMMONWEALTH 2, 2:15 P.M. TO 3:15 P.M.

Session 3pSC

Speech Communication: Celebration of Kenneth N. Stevens' Contributions to Speech Communication: Past, Present, Future II

Helen Hanson, Cochair
ECE Dept., Union College, 807 Union St., Schenectady, NY 12308

Stefanie Shattuck-Hufnagel, Cochair
MIT, 36-511 MIT, 77 Mass Ave, Cambridge, MA 02139

Open Microphone Session

Plenary Session and Awards Ceremony

Judy R. Dubno,
President, Acoustical Society of America

Presentation of Certificates to New Fellows

Robert D. Celmer – For contributions to undergraduate education in acoustics

Frederick Gallun – For contributions to cognitive issues in hearing impairment

Kirill V. Horoshenkov – For contributions to outdoor sound propagation, remote sensing, and acoustics of porous materials

Jeffrey A. Ketterling – For contributions to cavitation luminescence and very high frequency sensing and imaging

Jennifer J. Lentz – For contributions on hearing loss and the perception of complex sound

Bart Lipkens – For contributions to undergraduate education and practical applications of nonlinear acoustics

Arlene C. Neuman – For contributions to classroom acoustics and hearing aid development

Sunil Puria – For contributions to middle- and inner-ear biomechanics and their practical applications

Purnima Ratilal – For contributions to bioacoustics and underwater acoustic scattering and reverberation

Juan Tu – For contributions to the physical acoustics of ultrasound contrast agents

Introduction of Award Recipients and Presentation of Awards

Aaron Moberly, 2015 Research Grant in Speech Science of the
American Speech Language and Hearing Foundation

Lily M. Wang, 2015 Student Council Mentoring Award

Richard Ruby, John Larson, and Paul Bradley, The American Institute of Physics Prize
for Industrial Applications of Physics

Laurel H. Carney, William and Christine Hartmann Prize in Auditory Neuroscience

Karim S. Sabra, Medwin Prize in Acoustical Oceanography

Matthew W. Urban, R. Bruce Lindsay Award

Henry Cox, Helmholtz-Rayleigh Interdisciplinary Silver Medal

Gerhard M. Sessler, Gold Medal

Barbara G. Shinn-Cunningham, Vice President's Gavel

Judy R. Dubno, President's Tuning Fork

ACOUSTICAL SOCIETY OF AMERICA

R. BRUCE LINDSAY AWARD

Matthew W. Urban

2015

The R. Bruce Lindsay Award (formerly the Biennial Award) is presented in the Spring to a member of the Society who is under 35 years of age on 1 January of the year of the Award and who, during a period of two or more years immediately preceding the award, has been active in the affairs of the Society and has contributed substantially, through published papers, to the advancement of theoretical or applied acoustics, or both. The award was presented biennially until 1986. It is now an annual award.

PREVIOUS RECIPIENTS

Richard H. Bolt	1942
Leo L. Beranek	1944
Vincent Salmon	1946
Isadore Rudnick	1948
J. C. R. Licklider	1950
Osman K. Mawardi	1952
Uno Ingard	1954
Ernest Yeager	1956
Ira J. Hirsh	1956
Bruce P. Bogert	1958
Ira Dyer	1960
Alan Powell	1962
Tony F. W. Embleton	1964
David M. Green	1966
Emmanuel P. Papadakis	1968
Logan E. Hargrove	1970
Robert D. Finch	1972
Lawrence R. Rabiner	1974
Robert E. Apfel	1976
Henry E. Bass	1978
Peter H. Rogers	1980
Ralph N. Baer	1982
Peter N. Mikhalevsky	1984
William E. Cooper	1986
Ilene J. Busch-Vishniac	1987
Gilles A. Daigle	1988
Mark F. Hamilton	1989
Thomas J. Hofler	1990
Yves H. Berthelot	1991
Joseph M. Cuschieri	1991
Anthony A. Atchley	1992
Michael D. Collins	1993
Robert P. Carlyon	1994
Beverly A. Wright	1995
Victor W. Sparrow	1996
D. Keith Wilson	1997
Robert L. Clark	1998
Paul E. Barbone	1999
Robin O. Cleveland	2000
Andrew J. Oxenham	2001
James J. Finneran	2002
Thomas J. Royston	2002
Dani Byrd	2003
Michael R. Bailey	2004
Lily M. Wang	2005
Purnima Ratilal	2006
Dorian S. Houser	2007
Tyrone M. Porter	2008
Kelly J. Benoit-Bird	2009
Kent L. Gee	2010
Karim G. Sabra	2011
Constantin-C. Coussios	2012
Eleanor P. J. Stride	2013
Matthew J. Goupell	2014

ENCOMIUM FOR MATTHEW W. URBAN

. . . for contributions to ultrasonic measurement of viscoelastic properties of tissues for biomedical applications

PITTSBURGH, PENNSYLVANIA • 20 MAY 2015

Matthew Urban received his Bachelor's degree in Electrical Engineering with a Spanish minor at South Dakota State University in Brookings, South Dakota. Matt wanted to enter the field of engineering after high school because it involved two things he loved: math and physics. While still an undergraduate, he became a Summer Undergraduate Research Fellow in the Mayo Clinic Biomechanics Laboratory, where he designed a multi-axial testing machine for cadaveric and prosthetic specimens.

Matt decided to attend graduate school at the Mayo Clinic in Rochester, Minnesota because of the many opportunities in research that were available at that institution. He chose to do his Ph.D. thesis on specimens and volunteers, more befitting medical research. It was also a benefit that Matt was working in the Midwest, close to his and his wife's family in South Dakota where he met his wife Joleen, a nurse, at the South Dakota State University They now have three children.

Matt began his academic investigations by studying the phase and vibration frequency of spheres in a viscoelastic medium, which resulted in a laboratory standard for measuring the mechanical properties of materials in the lab. Subsequently, he has contributed greatly to the field of radiation force shear wave dispersion measurements by developing a phase aberration correction method that used the motion in the tissue from the ultrasound radiation force to correct for phase aberration in the transmitting beam. In addition, Matt has been a key investigator in the Mayo Clinic Ultrasound Research Laboratory in investigating vibrometry of the myocardial wall. In 2009 he reported, along with colleagues, methods for using dispersion of shear wave speed for estimating tissue elasticity and viscosity. In 2010 Matt reported a method of modulating ultrasound for producing multi-frequency radiation force to make measurements at multiple frequencies of the mechanical properties of tissue. In addition he was the principal mentor for a graduate student whose paper on material property estimation for tubes and arteries using the radiation force method resulted in a significant amount of work in which modes of vibration of *in vivo* arteries have been studied. During these investigations, it was discovered that the propagation of shear waves in the wall of the heart obeyed the Lamb wave dispersion rules and Matt, along with others, was key in helping to work out those relationships.

Matt has also been directly responsible for implementing vibro-acoustography on a clinical ultrasound system and helping to evaluate the system. Recently, Matt has been closely involved with the development of a completely new method of measuring two-dimensional distributions of mechanical properties of tissue, which is called the comb-push ultrasound shear elastography method. He has also been working on methods for measuring the viscoelastic properties of the bladder wall in patients with compromised bladder function. Overall, these investigations have led to new knowledge about the relationships between the mechanical properties of tissue and propagating shear waves. Matt's investigations into dispersion diagrams for propagating shear waves has led to completely new methods for measuring the attenuation of shear waves in tissue using model free analyses. He has done a significant amount of work developing vibro-acoustography as implemented on a General Electric machine now in clinical investigation. Matt has published 56 papers in well-respected peer reviewed journals, including as first author of a review paper in *Current Medical Imaging Reviews* (2012) on the subject of vibro-acoustography and its applications.

Matt directed a sponsored research program from a large ultrasound imaging company in which some of the fundamental new methods that he and others have developed are being investigated for possible inclusion into their commercial instruments.

Matt has a Research Project Grant (R01) from the National Institute of Diabetes and Digestive and Kidney Diseases at the National Institutes of Health on the subject of measurements of mechanical properties of the anisotropic cortex of the kidney after transplantation in humans. Matt's current research includes investigation of shear wave methods in transplanted kidneys as funded by his R01, and in addition, he is studying special beams and their beam properties for use in shear wave elastography measurements as funded by an international ultrasound manufacturer.

Matt has received several awards from the Acoustical Society of America (ASA), including best student paper awards in 2006 and 2007 and young investigator travel grants. He is a member of the ASA Technical Committee on Biomedical Acoustics and co-organized special sessions in 2010 and 2012. Matt is a member of the American Institute of Ultrasound in Medicine (AIUM) and a senior member of the Institute of Electrical and Electronics Engineers (IEEE). He has reviewed for many of the top journals in the field of ultrasonic imaging and biomedical engineering and is also currently supervising two graduate students.

Matt is an outstanding teacher, a creative, highly-capable investigator in the field of ultrasound measurements of mechanical properties of tissue. His exceptional legacy of development and innovation in the field makes him an outstanding recipient of the Lindsay award. We are delighted to congratulate Matt on behalf of his many colleagues, family, and friends around the world for being awarded the R. Bruce Lindsay Award of the Acoustical Society of America.

JAMES F. GREENLEAF

ACOUSTICAL SOCIETY OF AMERICA
HELMHOLTZ-RAYLEIGH INTERDISCIPLINARY
SILVER MEDAL
in

Signal Processing in Acoustics, Underwater Acoustics, and Engineering Acoustics

Henry Cox

2015

The Silver Medal is presented to individuals, without age limitation, for contributions to the advancement of science, engineering, or human welfare through the application of acoustic principles, or through research accomplishment in acoustics.

PREVIOUS RECIPIENTS

Helmholtz-Rayleigh Interdisciplinary Silver Medal

Gerhard M. Sessler	1997
David E. Weston	1998
Jens P. Blauert	1999
Lawrence A. Crum	2000
William M. Hartmann	2001
Arthur B. Baggeroer	2002
David Lubman	2004
Gilles A. Daigle	2005
Mathias Fink	2006
Edwin L. Carstensen	2007
James V. Candy	2008
Ronald A. Roy	2010
James E. Barger	2011
Timothy J. Leighton	2013
Mark F. Hamilton	2014

Interdisciplinary Silver Medal

Eugen J. Skudrzyk	1983
Wesley L. Nyborg	1990
W. Dixon Ward	1991
Victor C. Anderson	1992
Steven L. Garrett	1993

ENCOMIUM FOR HENRY COX

. . . for fundamental and practical contributions to array signal processing, underwater acoustics, and sonar systems engineering

PITTSBURGH, PENNSYLVANIA • 20 MAY 2015

Henry Cox's career has been a remarkable demonstration of versatility. He has met the highest standards of scholarly achievement in his contributions to acoustic signal processing and underwater acoustics. He also served with great distinction as a career navy officer managing major programs that had a critical impact on national security during the Cold War era. The ability to serve simultaneously in both these roles with such success is very rare.

Harry, as he is known by his colleagues, attended the College of the Holy Cross where he graduated cum laude prior to entering the Navy. He received his doctorate in electrical engineering from the Massachusetts Institute of Technology while serving in the Navy. He was granted a sabbatical, which he spent at Scripps Marine Physical Laboratory. During that time he interacted with prominent acousticians and signal processors and wrote several papers that are often cited as landmark papers in underwater acoustic signal processing. One of those papers is his 1973 paper published in the *Journal of the Acoustical Society of America* (JASA) on the spatial correlation of noise in the ocean. In this paper, realistic noise fields are described by a rigorous mathematical presentation, providing a foundation for analysis of adaptive processing methods.

He subsequently served in a number of important program management roles as a naval officer, including the undersea surveillance program, prior to retiring from the Navy and joining Bolt, Beranek and Newman Inc. (BBN) as a divisional vice president. After a number of years at BBN, he left to join ORINCON, and is currently a Senior Fellow at Lockheed Martin. Throughout his career, he has contributed important papers that are major influences on how we think about acoustic signal processing. His work is widely cited and he is a Fellow of the Acoustical Society of America (ASA) and the Institute of Electrical and Electronics Engineers (IEEE), and a member of the National Academy of Engineering. He continues to make important contributions through both his own work and that of the cadre of young scientists and engineers at multiple labs for whom he serves as mentor.

As a Navy program manager, CAPT H. Cox, USN, set very high standards for technical rigor and had a reputation for holding others to them. One can imagine the surprise of a young researcher, armed with briefing materials appropriate for most officers, upon finding that his audience knew more about the subject than he did. Harry also had the broad vision necessary to understand and guide major programs, and he accepted and exercised the responsibility to ensure that government funds were spent wisely. During his Navy career, he was able to save the government substantial sums of money by avoiding poorly considered approaches to sonar system development in favor of well-conceived and technically sound approaches. It is altogether too rare to encounter a program manager in the government today with both the technical and managerial strength to do the job as well as Harry did.

Harry's most important contributions are in his technical work in the areas of signal processing applied to sonar systems. His broad knowledge of propagation and noise mechanisms in the ocean underlies his ability to understand how the ocean acoustic environment impacts sonar systems, and to then develop signal processing methods that are robust in the face of environmental factors that may differ from assumptions, may not be well characterized, and may be variable. Harry is interested in taking advantage of sophisticated adaptive methods to optimize the processor, but he is wary of problems that often arise due to mismatch between the real ocean acoustics and the assumptions built into the adaptive processor. Among his most influential contributions is a paper entitled "Robust Adaptive Beamforming" published in 1987 in the IEEE Transactions on Acoustics, Speech, and Signal Processing. In this paper he develops a practical approach to using a white noise gain constraint to mitigate mismatch problems in a minimum variance distortionless response beamformer. This paper has stimulated much progress in practical implementations of adaptive methods in sonar, and has led to a variety of implementations that significantly improve the capabilities of Navy systems in use today. Harry's career is characterized by many such contributions. He has an excellent grasp, intuitive one might say, of the mathematical foundations of signal processing as well as the physics of sound in the ocean and is able to combine them in a way that provides practical solutions to difficult problems under stressful ocean conditions. Major contributions include rigorous development of mathematical descriptions of realistic ocean noise fields, application of multi-rate adaptive processing methods to cancel out rapidly moving interferers, establishing fundamental principles for design of bistatic active sonar systems, advanced methods for using Doppler processing in active sonar, approximate ray angle diagrams for quickly assessing propagation effects on the vertical distribution of signal and noise fields in the ocean, and simplified techniques for matched field processing methods.

The insights Harry has provided have served as guidelines and basic rules of thumb for many other scientists, engineers, and students involved in development of sonar and signal processing technologies. And speaking of rules of thumb, we can't help but note one of Harry's hallmarks, his back-of-the-envelope rules, formulas, intuitions and approximations—some almost more precise than an actual calculation—that he tosses off like confetti. It's an impressive tour de force. Harry is well known and appreciated for his ability and willingness to guide others, particularly young scientists and engineers, and give them the benefit of his experience and his extensive technical knowledge. Many have benefitted from knowing him, from learning something new or finding a better way to analyze a problem and from understanding that while technical correctness is necessary, without honest interpretation it is insufficient. He has been a mentor to multiple generations of underwater acousticians and signal processors and he continues in this role today. We look forward to his continuing technical contributions, and his wise advice and counsel. For the reasons cited here, we are pleased to present Harry Cox with the Helmholtz-Rayleigh Interdisciplinary Silver Medal.

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ACOUSTICAL SOCIETY OF AMERICA

GOLD MEDAL

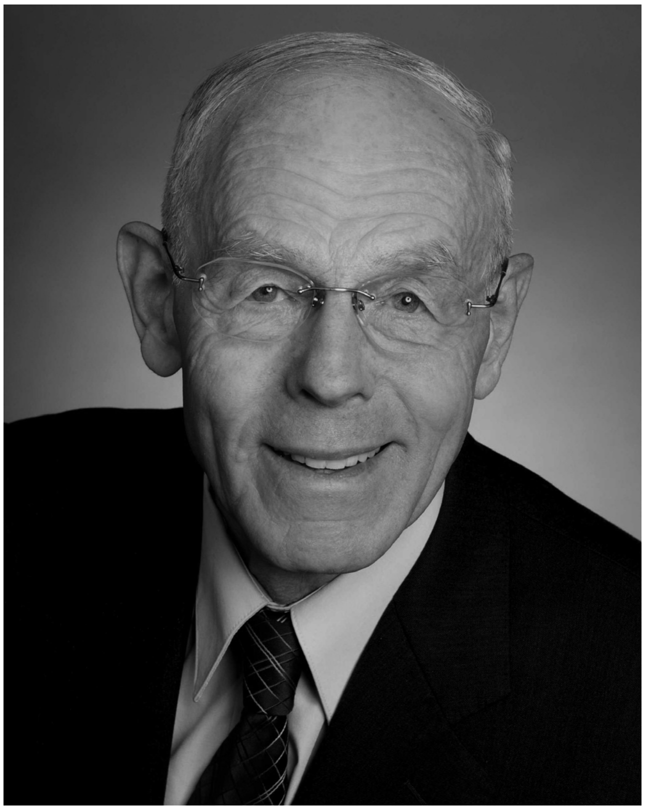
Gerhard M. Sessler

2015

The Gold Medal is presented in the spring to a member of the Society, without age limitation, for contributions to acoustics. The first Gold Medal was presented in 1954 on the occasion of the Society's Twenty-Fifth Anniversary Celebration and biennially until 1981. It is now an annual award.

PREVIOUS RECIPIENTS

Wallace Waterfall	1954
Floyd A. Firestone	1955
Harvey Fletcher	1957
Edward C. Wente	1959
Georg von Békésy	1961
R. Bruce Lindsay	1963
Hallowell Davis	1965
Vern O. Knudsen	1967
Frederick V. Hunt	1969
Warren P. Mason	1971
Philip M. Morse	1973
Leo L. Beranek	1975
Raymond W. B. Stephens	1977
Richard H. Bolt	1979
Harry F. Olson	1981
Isadore Rudnick	1982
Martin Greenspan	1983
Robert T. Beyer	1984
Laurence Batchelder	1985
James L. Flanagan	1986
Cyril M. Harris	1987
Arthur H. Benade	1988
Richard K. Cook	1988
Lothar W. Cremer	1989
Eugen J. Skudrzyk	1990
Manfred R. Schroeder	1991
Ira J. Hirsh	1992
David T. Blackstock	1993
David M. Green	1994
Kenneth N. Stevens	1995
Ira Dyer	1996
K. Uno Ingard	1997
Floyd Dunn	1998
Henning E. von Gierke	1999
Murray Strasberg	2000
Herman Medwin	2001
Robert E. Apfel	2002
Tony F. W. Embleton	2002
Richard H. Lyon	2003
Chester M. McKinney	2004
Allan D. Pierce	2005
James E. West	2006
Katherine S. Harris	2007
Patricia K. Kuhl	2008
Thomas D. Rossing	2009
Jiri Tichy	2010
Eric E. Ungar	2011
William A. Kuperman	2012
Lawrence A. Crum	2013
Brian C. J. Moore	2014



ENCOMIUM FOR GERHARD M. SESSLER

. . . for the development of electret and silicon-based micromachined microphones

PITTSBURGH, PENNSYLVANIA • 20 MAY 2015

Gerhard M. Sessler received his Ph.D. from the University of Göttingen in 1959 under the supervision of Professor Erwin Meyer. Upon graduation, he moved to the United States where he joined the technical staff in the Acoustics Research Department at Bell Laboratories. There he worked on acoustic transducers, room acoustics, and studies of transducer materials until 1975 when he returned to academia in Germany. He served as Professor of Electrical Engineering at the Darmstadt University of Technology from 1975 to 1999 and as Professor emeritus from 1999 to the present. He continues to be active in research.

At Bell Labs, Gerhard's major contribution was the development of the electret microphone in the early 1960's, together with James E. West. The electret microphone has dominated the microphone market for over 40 years, with recent production rates of more than 2 billion per year, for applications ranging from toys to cell phones to professional studio and measuring microphones. Electret microphones allowed, for the first time, practical directional arrays ranging from a simple differential transducer to broadband higher-order arrays with toroidal and other polar characteristics. Companies such as Brüel & Kjær utilize electret transducers for high precision measuring microphones with good thermal stability.

The research that led to the electret microphone forged lifelong personal relationships that resulted in collaborations leading to many new concepts in transducers, room acoustics, physical acoustics, electroactive transducer materials, and atmospheric acoustics. These included measuring the very low frequency sound generated from rockets launched in Florida that reached Murray Hill, New Jersey after travelling through the atmosphere for a distance of about a thousand miles.

Despite the dominance of electret microphones, Gerhard realized that even smaller transducers would be necessary, as communications devices were destined to become smaller. He also realized that techniques similar to those used for integrated circuit production could be used to make transducers that had the potential to become the microphones of the future. With this vision he began a rigorous program at the Technical University of Darmstadt on silicon based transducers using techniques developed for micro-electro-mechanical-systems (MEMS) sensors. Gerhard and his students were the first to introduce an integrated silicon condenser microphone (1983), and later he showed that the small air gap typical for these transducers drastically reduced the required bias voltage to levels available in integrated circuit systems. MEMS microphones are now in production and have begun to replace electret microphones in cell phones and other applications where space is restricted and the matching of microphones for array processing is important. Today, more than a billion units per year are produced based on the principles first published by Gerhard and his students.

Professor Sessler is a successful university professor. All of his former Ph.D. students are working in acoustics and three are professors at universities in Germany. One graduate student, Jens Meyer, together with Gary Elko, invented a small spherical microphone array based on new technology that drastically reduced both the size and signal processing required to form directional beams. They further developed this technology and later formed a company to produce a spherical array known as the Eigenmike.

Recently, Gerhard investigated cellular polypropylene films poled by breakdown charging the air voids resulting in a charge separation within the voids resembling dipoles. The poled films, also referred to as ferroelectrets, have piezoelectric coefficients higher by an order magnitude than found in other polymer materials. These high values make this material ideal for flat panel loudspeakers, with applications in mobile and smart phones, but also for microphones, ultrasonic, and electromechanical transducers, and for energy harvesters.

Gerhard Sessler also contributed to the introduction of digital signal processing for the evaluation of small and large auditoriums and concert halls. These methods were first used for the evaluation of the acoustics of Philharmonic Hall in New York, directed by Manfred R. Schroeder. Later on, together with Jim West, he discovered what is known as the seat dip effect, the frequency dependent attenuation due to the spacing between seat rows, and improved knowledge influencing concert hall design.

It is important to note that Gerhard is not only an engineer inventing new technical devices, he is also a careful physicist who cares for the basic understanding and the physics behind his inventions. For example, he systematically examined the charge storing mechanisms in many electret materials and piezopolymers as well as in the new ferroelectrets. He is author of more than 150 articles in peer-reviewed journals including 21 in the *Journal of the Acoustical Society of America*, and of several textbooks on acoustics and charge storage in polymers. His books on charging effects, distributed all over the world, have become the basic textbooks on this topic. He was also co-author with Ning Xiang of a memorial volume published in 2014 in honor of Manfred R. Schroeder, his friend and collaborator at Bell Laboratories. Gerhard holds over 100 international patents and 21 U.S. patents. The first one on the electret microphone with James E. West, was issued on January 14, 1964.

Gerhard Sessler is Fellow of the Acoustical Society of America, the Institute of Electrical and Electronics Engineers, and the American Physical Society. His professional achievements have been honored by numerous prestigious scientific awards including the Callinan Award of the Electrochemical Society, the Senior Award of the IEEE Group on Audio and Electroacoustics, the Thomas W. Dakin Award of the IEEE Dielectrics and Electrical Insulation Society, the Helmholtz-Rayleigh Interdisciplinary Silver Medal of the Acoustical Society of America, the Benjamin Franklin Medal in Electrical Engineering of the Franklin Institute, and the IEEE Maxwell Award (IEEE and Royal Society of Edinburgh), several of these together with Jim West. He was inducted into the US National Inventors Hall of Fame in 1999.

Gerhard Sessler has devoted most of his adult life exploring new ideas in acoustics, and undoubtedly he will continue contributing to the advancement of this field of science. In recognition of his extraordinary contributions to acoustics, we congratulate Gerhard M. Sessler for being awarded the Gold Medal of the Acoustical Society of America.

REINHARD LERCH
JAMES E.WEST

Session 4aAAa

Architectural Acoustics: Architectural Acoustics Potpourri

Ana M. Jaramillo, Cochair
Ahnert Feistel Media Group, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444

Christopher L. Barnobi, Cochair
CSTI Acoustics, 16155 Park Row Suite 150, Suite 101, Houston, TX 77084

Contributed Papers

8:00

4aAAa1. Comparisons between the preferences of musicians and non-musicians in response to varying room acoustics using two different testing methods. Martin S. Lawless and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, msl224@psu.edu)

Although the brains of musicians and non-musicians differ both on an anatomical and functional level, these groups tend to experience similar emotions in response to the same, unaltered musical excerpts (Bigand/Poulin-Charronnat *Cognition* 2006). Contrarily, preliminary results of the authors and colleagues suggest that the two groups have diverse preferences when presented with the same excerpt convolved with different room conditions. A more comprehensive investigation was conducted to determine differences in preference between musicians and non-musicians in response to stimuli of with different reverberation times, ranging from anechoic to very reverberant. Room acoustics models were used to create auralizations for a number of motifs. Subjects initially rated the stimuli in terms of overall preference, followed by rating their perception of reverberance. Two distinct subjective testing methods, successive and comparative, were utilized and compared. The successive method required the participants to rate each stimulus separately in succession, while the comparative method allowed the subjects to compare and rate each stimulus within a set with the rating scale for each stimulus on one screen. The comparison between methods was performed to validate future testing conducted with the same stimuli in more constrained settings, specifically in a functional magnetic resonance imaging (fMRI) scanner.

8:15

4aAAa2. Investigating the effect of arrival time of diffuse reflections on listener envelopment. Brandon Cudequest, M. Torben Pastore, and Jonas Braasch (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, bcudequest@gmail.com)

Listener Envelopment (LEV) is a quality of diffuse sound fields, and a highly sought after attribute of performance venues. However, the ability for an enclosure to achieve ideal, diffuse sound field is often difficult, and varies from space to space. Thus, a rigid transition time between early and late energy is an insufficient way of evaluating LEV. Current theories on listener envelopment focus on the arrival time of the reflections within the impulse response, but typically disregard the diffusivity of these reflections, building on the fact that the impulse response generally becomes more diffuse over time. An alternative model is proposed, where the listener envelopment is determined by both the diffusivity and the arrival time of reflections. The model disregards the current 80-ms criterion and also allows reflections earlier then this to contribute to LEV. A 64-channel wave field synthesis system is used to perceptually evaluate the effects of spatially and temporally diffuse sound components as a function of arrival time.

8:30

4aAAa3. Connecting the sense of envelopment to specific components of the sound field using perceptually motivated auralizations. Matthew T. Neal and Miche C. Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mtn5048@psu.edu)

Envelopment is known to be a key attribute related to overall room impression. Despite this importance, limited research has been done to identify the specific components of the sound field that contribute to envelopment. The goal of this study was to determine the timing and spatial distribution of reflections contributing to envelopment. A subjective study was conducted using a range of simulated auralizations, which were played back over a three-dimensional loudspeaker array in an anechoic chamber. For each auralization, subjects rated their perceived envelopment. A real-time acoustic simulation program was developed in Max to generate the signals, which simulated early sound with the image-source method and late sound with statistical reverberation. When creating the stimuli, the program produced immediate auditory feedback in response to adjusting the input parameters. The signals were quantified through impulse response measurements to ensure a wide range of conditions. The subjects' envelopment ratings were correlated to different components of the sound field, to evaluate how specific arrival time of reflections and spatial characteristics contribute to envelopment. These results could possibly be used to determine the effectiveness of existing envelopment metrics and potentially contribute to developing a new measure to predict envelopment. [Work supported by NSF Award-1302741.]

8:45

4aAAa4. Image source model for small room acoustics. Ambika Bhatta, Charles Thompson, and Kavitha Chandra (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, ambika_bhatta@student.uml.edu)

In this investigation, the impulse response obtained using exact solution comprising of the Inverse Laplace transform (ILT) of the direction cosine of the incident angle is undertaken. The results are compared to those obtained by Allen and Berkley (JASA **65**, 943–950, 1979). In addition, frequency dependent wall impedance effects are considered. Numerical efficiency and accuracy of the image solution are evaluated.

9:00

4aAAa5. Marketing architectural acoustics to non-acousticians (a.k.a. the unwashed masses). Samuel V. Diaquila (Architecture, Kent State Univ., 21863 Aurora Rd., Cleveland, OH 44146, sam@baswaphon.com)

Architects understand aesthetic issues: color, texture, glare, serenity, excitement, as well as mechanical issues; fire suppression, air changes, and restroom requirements. Their thought process rarely considers the intangible world of acoustics. Their clients (the "owner") is even further removed. In

4a THU. AM

today's construction environment, how does the acoustical consultant market his or her services successfully to decision makers who are faced with ever decreasing budgets and an ever increasing bombardment of information? Even more challenging is ensuring that the acoustical consultant's recommendations are implemented. Learn ways to awaken your potential client's sensitivity and awareness of acoustics. Changing your sales presentation from a technical (acoustical consultants are generally "blind"), relying on computer generated acoustical reports that resemble mathematical thesis papers to an emotion based technique (potential clients are generally "deaf"), that brings the science of acoustics to life at a tangible human level. Everyone reacts to bad acoustics, with the exception of concert halls and restaurants, acoustics is rarely discussed and often not one of the end user's priorities until there is a problem needing a retrofit.

9:15

4aAAa6. The resonance of tapering spiral chambers. Paula Pino (Paulapart, 142 Irving Ave., Apt. 1R, Brooklyn, NY 11237, paul@paulapart.com)

This paper will explore the acoustical resonance of several cochlea-inspired sculptures (i.e., spiraling, tapering forms) made of glass, ceramics, and other materials. Each sculpture will function as an acoustic chamber and will be equipped with a loudspeaker and a "mouth" through which sound will pour out. Using frequency sweeps, feedback loops, reverb convolution, and acoustic prediction software I will measure the resonant and reverberant responses of these acoustic sculptures and compare them with analogous forms (e.g., animal horns, brass instruments, and mammalian cochlea). Special focus will be on the fundamental resonant frequencies and bass response of each shape. Extrapolation with these data will inform the production of larger, more architectural acoustic chambers modeled with similar spiraling geometry.

9:30

4aAAa7. Architectural acoustics illustrated. Michael Ermann (Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu)

Novel discoveries, non-obvious findings, and counter-intuitive conclusions will be excerpted and highlighted from the just-released book *Architectural Acoustics Illustrated* (authored by the presenter, Wiley, 2015). The text filters the content of architectural acoustics through the graphic and built language of architecture. In writing the book, building material choices, spatial relationships, best-practices, and data were explored through drawing and animated videos. Summaries of the findings will be demonstrated graphically to establish relationships in sound absorption, room acoustics, sound isolation, and noise control.

9:45–10:00 Break

10:00

4aAAa8. Simulation and auralization of a concert hall's inhomogeneous sound field using finite difference time-domain methods and wave field synthesis. Kelsey Hochgraf, Jonathan Botts, Ning Xiang, and Jonas Braasch (Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, hochgk@rpi.edu)

Auralization methods have been used for a long time to simulate the acoustics of a concert hall for different seat positions. The goal of the research project presented here is to apply the concept of auralization to a larger audience area that the listener can walk through to compare differences in acoustics for a wide range of seat positions. For this purpose, the low frequency acoustics of Rensselaer's Experimental Media and Performing Arts Center (EMPAC) Concert Hall are simulated using finite difference time-domain (FDTD) methods to create signals for a 128-channel wave field synthesis (WFS) system located at Rensselaer's Collaborative Research Augmented Immersive Virtual Environment (CRAIVE) Laboratory. By allowing multiple subjects to dynamically experience the concert hall's acoustics at the same time, this research gains perspective on what is important for achieving objective accuracy and subjective plausibility in a simulation and auralization. Efforts are made to maintain efficiency of wave-based modeling, and methods for evaluating the final auralization are explored from both objective and perceptual standpoints.

10:15

4aAAa9. Practical desktop full-wave architectural acoustic solutions. Patrick Murray, Jeff Cipolla, and Adam Hapij (Appl. Sci., Weidlinger Assoc., 40 Wall St., FL19, New York, NY 10005, patrick.murray@wai.com)

An explicit, time-domain, finite-element method is shown to calculate the reverberation time in realistic acoustic spaces using desktop computing resources. The reverberation time is defined as the time required for reflections of direct sound to decay 60 dB, and is also the principal quantity in architectural acoustics across the frequency ranges of interest. Current industry practice for calculating the reverberation time involves empirically derived formulae which cannot account for the architectural complexity of modern acoustic spaces or the detailed placement of acoustic treatments. Computing advances over the past several years have made it possible to calculate the reverberation time using finite element models in the time domain. Explicit finite-element codes are distinct from traditional ones because they operate by integrating the governing equations of mass, momentum, and energy in the time domain, avoiding the need for matrix generation, storage, and factorization. The feasibility of this full-wave approach is demonstrated using the configurations of real example acoustic spaces. Comparison is made with empirical calculations and experimental measurements. It is also shown how the enhancements can be made to a space by the addition of acoustic damping material.

10:30

4aAAa10. Beyond ISO3382—Measuring acoustics with live sound. David H. Griesinger (Res., David Griesinger Acoust., 221 Mt. Auburn St #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Current acoustic measurements techniques typically require time and heavy equipment. The effort and expense would be justified if the results accurately predicted the sound quality from a performance at an individual seat, but they do not. If one believes that it is possible to hear and accurately report sound quality from live sound in different seats, then it must be possible to measure it this way. This paper describes a model of human hearing that promises to provide this ability, at least for quality aspects relating to clarity. The model is based on the ease with which information is carried from one or more sources to a listener. For speech, this involves both reverberant masking from late reverberation, and interference to the direct sound from early reflections. To make such a measurement, we need to model how the ear and brain system precisely localizes sound sources, and separates their sound from each other and acoustic interference without added information from context, grammar, or prior knowledge. A description of such a model will be presented, along with results obtained from live sound.

Session 4aAAb

Architectural Acoustics and Education in Acoustics: Up and Coming Architectural Acousticians: Past Student Paper Award Winners Report

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

Invited Papers

9:45

4aAAb1. Acoustic adventures over continents, among cultures: Under tides of academia and profession. Zühre Sü Gül (MEZZO Studyo LTD, METU Technopolis KOSGEB-TEKMER No112, ODTU Cankaya, Ankara 06800, Turkey, zuhre@mezzostudyo.com)

After obtaining a Bachelor of Architecture and an MFA with an emphasis in amphitheater acoustics, I was accepted into Rensselaer Polytechnic Institute's unique Graduate Program in Architectural Acoustics. There I completed an MS degree focusing on architectural acoustics in coupled spaces and was honored to receive the Robert Newman Award, a TCAA Student Paper Award and an ASA commendation for student design. Since 2006 I have applied my academic focus to design projects, first at RBDG Inc. in Dallas, and later at home in Turkey at MEZZO Stüdyo, the firm I co-founded in 2009. Turkish governmental supports have afforded MEZZO the opportunity to become a critically recognized "R&D incubator." We've also completed over one hundred design projects, both in-house and teamed with architectural-engineering groups around the world and are proud members of NCAC (National Council of Acoustical Consultants). During the restricted times of not managing my firm—and my 3-year-old son!—I enjoy giving university lectures to inspire future generations about exciting diversities in the field of acoustics. Luckily, I just completed my PhD degree on Architectural Acoustics again, and I am much proud and happy to share this whole adventure.

10:00

4aAAb2. Strategies for guiding undergraduate acoustics students through independent research projects. Lauren M. Ronsse (Audio Arts and Acoust., Columbia College Chicago, 33 E. Congress Pkwy, Ste. 601, Chicago, IL 60605, lronsse@colum.edu)

Students in the B.S. Acoustics Program at Columbia College Chicago must successfully complete an independent research project as part of the curriculum. Since this is the first independent research experience for most of the students, measures are taken to guide them through the research process. Also, an electronic research notebook tool has been developed, providing a structured format for the students to document their research. The teaching techniques employed to prepare and guide the undergraduate acoustics students through the research process will be described. Recent student research projects that utilized the electronic research notebook reporting format will also be highlighted.

10:15

4aAAb3. Cultureshock Los Angeles: A midwesterner's guide to acoustics, food trucks, and surfing. Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, srawlings@veneklasen.com)

Samantha Rawlings & Joshua Magee were awarded a Student Paper Award at the Fall 2007 Acoustical Society of America conference in New Orleans for undergraduate research on sound radiation from footfall noise. This student project paved the way for Samantha to move to Los Angeles to join Veneklasen Associates, which was a wholly un-looked for opportunity for someone who once laughed outright at a Hardees commercial showing a hamburger wrapped in lettuce. Since moving to LA, Samantha has become a LEED AP and Project Manager and has had the chance to work on projects of all size and scope; everything from the "mystery noise" cold call to landmark project such as City Creek Center in Salt Lake City and Wilshire Grand in downtown Los Angeles. Additionally, Samantha has participated in research projects that have been presented at ASA, NoiseCon, and Internoise conferences. In this presentation, Samantha reflects on the surprising events of the past seven years and the adventure of learning to fit in and love Southern California.

10:30

4aAAb4. Parallels in scattering research between architectural and underwater acoustics. Derek R. Olson (Appl. Res. Lab., The Penn State Univ., P.O. Box 30, State College, PA 16804, dro131@psu.edu)

Acoustical scattering is a central area of research in the fields of underwater and architectural acoustics. Although the mathematical formulation of the scattering problem is common to both fields, their goals and environments have motivated divergent analysis techniques, and quantities associated with the pressure field due to scattering. One of the primary differences is the role of statistical averaging.

Ensemble-averaged quantities can often be used in ocean acoustics because the seafloor may have statistically homogeneous roughness and geoacoustic properties over large areas. In the built environment, the acoustic field can interact many times with a single acoustic diffusor, and deterministic properties of the scattered field are of greater import. Acoustical quantities such as the scattering cross section, scattering coefficient, and the diffusion coefficient will be defined and compared. Examples will be given that highlight cases when ensemble-averaged quantities can aid or hinder design of acoustical diffusors, depending on the design constraints.

10:45

4aAAb5. Effects of conductor verbal and nonverbal instructions on changes in singer voicing behaviors: A survey of recent research. Jeremy N. Manternach (College of Education/School of Music, The Univ. of Iowa, 311 Communications Ctr., Iowa City, IA 52242, jeremy-manternach@uiowa.edu)

Vocal and choral music professionals seek to guide their singers to efficient and beautiful vocal production. They do so using a multitude of techniques, some of which have been passed along from one generation of teachers to the next. These techniques include giving singers a particular focus of attention and displaying varied conducting gestures designed to evoke specific voicing behaviors. Despite the prevalence of these techniques, few researchers have investigated whether they consistently change the acoustical output or the muscle engagement used by individual singers or choristers. This presentation provides an overview of recent research that has sought to quantify the voicing behaviors of individual singers as they (a) sang a short melody while focusing on various internal or external foci, (b) viewed varied conductor preparatory (inhalation) gestures prior to singing a short melody, and (c) viewed varied conductor final release (cutoff) gestures at the end of a short melody. The resulting acoustical, biomedical, and perceptual data inform the pedagogical implications that are discussed throughout.

11:00

4aAAb6. Continuing research endeavor in room acoustic effects on speech communication and psychoacoustics. Z. Ellen Peng (Inst. of Tech. Acoust., RWTH Aachen Univ., RWTH Aachen, Kopernikusstraße 5, Aachen, Germany, ellen.z.peng@gmail.com)

I received the best student paper award in architectural acoustics at the San Francisco ASA meeting in 2013. I presented the first half of my dissertation project on designing acoustics for linguistically diverse classrooms by investigating the effect of room acoustics on speech comprehension by native and non-native English-speaking listeners. In the following year, I continued to complete my dissertation and received my PhD in architectural engineering from the University of Nebraska-Lincoln. Beginning in 2015, I am a postdoctoral fellow at RWTH Aachen University in Germany as part of the iCARE (icareitn.eu) research program funded by a Marie-Curie Initial Training Network (ITN) grant from the European Union. My role is to help improve an existing electronic system in extending its application of virtual acoustics testing on children with hearing impairment. Upon receiving the best student paper award, I continue to enjoy multidisciplinary research that combines architectural acoustics, speech communication, and psychoacoustics.

11:15

4aAAb7. An approach to communicating in-field acoustical performance to architects as it relates to end user experience. Ari M. Lesser and Adam P. Wells (Acoustics, Cerami & Assoc., Inc., 404 Fifth Ave. 8th Fl., New York, NY 10018, alesser@ceramiassociates.com)

A study was conducted within an industry leading architecture firm to benchmark the noise isolation class, background noise level, and reverberation time throughout the firm's spaces. Testing was conducted in accordance with the general guidelines of ASTM E336-14, ANSI S1.13, and ASTM E2235-04. Acoustical test results were compared to employee subjective satisfaction survey responses in an effort to bridge the gap between acoustical design recommendations and end users' expectations and experiences of their environment. [Work supported by Cerami & Associates, Inc.]

11:30

4aAAb8. Navigating and developing acoustics research and curriculum within the Georgia Tech Center for Music Technology. Timothy Hsu (School of Music, Georgia Inst. of Technol., 840 McMillan St., Atlanta, GA 30332-0456, timothy.hsu@music.gatech.edu)

Acoustics at the Georgia Institute of Technology has primarily been based within mechanical engineering with various other faculty spread throughout the university. Within the last decade, the School of Music has developed an internationally recognized graduate degree program and research center in music technology. Since winning the Student Paper Award, my professional career has started in a unique faculty position within the School of Music at Georgia Tech, where my responsibilities are split between being an ensemble/choral conductor and an active member of the Center for Music Technology. One of the recent challenges is the development of an innovative undergraduate curriculum model that fuses music fundamentals, music technology, general musicianship, and engineering/computer science. At the graduate level, new coursework I have created has included a musical acoustics course for music technologists and a historical acoustics and modeling course. Additionally, developing a research track of acoustics within the confines of music technology has allowed my research to morph into musically related areas that my graduate research did not allow. Areas of current research include historical architectural acoustics, active temperament, synthesis, and noise control.

11:45

4aAAb9. Teaching acoustics to people with non-scientific backgrounds. Ana M. Jaramillo (Ahnert Feistel Media Group, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu)

With a background in architecture, the author has spent some time developing curriculum for the teaching of acoustics to architecture, design, and music technology majors. Acoustic software is often employed as a tool for the understanding of acoustic concepts, especially helpful when teaching people with non-scientific backgrounds. Through her work with Anhert Feistel Media Group (AFMG), the author has had a chance to take a closer look at acoustic software and spend some time understanding the needs of acoustic simulation software users and learners.

Session 4aAO

Acoustical Oceanography: General Topics in Acoustical Oceanography

Zoi-Heleni Michalopoulou, Chair
Mathematical Sciences, New Jersey Institute of Technology, 323 ML King Blvd., Newark, NJ 07102

Chair's Introduction—8:30

Contributed Papers

8:35

4aAO1. Fate of methane gas bubbles emitted from the seafloor along the Western Atlantic Margin as observed by active sonar. Liam Pillsbury and Thomas C. Weber (CCOM, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, LPillsbury@ccom.unh.edu)

A method for characterizing and quantifying marine methane gas seeps along the Western Atlantic Margin (U.S. East Coast) was developed and applied to 70 free-gas seeps observed by the R/V Okeanos Explorer in 2012 and 2013, in water depths ranging from 300–2000 m. Acoustic backscatter from an 18 kHz split-beam echo sounder and a 30 kHz multi-beam echo sounder provided information on the height to which the gas seeps rose from the seabed. Profiles of the depth-dependent target strength and scattering strength per unit depth were generated from the acoustic data. These profiles were compared to models of the evolution of rising bubbles in order to help constrain the ultimate fate of the bubbles. Of particular interest are comparisons of profiles for seeps originating below, at, and above the gas hydrate stability zone.

8:50

4aAO2. Diameter and density dependent target strength of submerged oil droplets measured by a broadband, high-frequency echo sounder. Scott Loranger (Earth Sci., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, sloranger@ccom.unh.edu) and Thomas C. Weber (Mech. Eng., Univ. of New Hampshire, Durham, NH)

Over two million tons of oil enters marine environments from anthropogenic sources annually with severe environmental consequences. The most effective method of cleaning spills is to biodegrade them by dispersing the oil as droplets. However, the ultimate fate of dispersed oil in the environment is largely unknown. Acoustic remote sensing may offer a means by which to assess the quantity, characteristics, and ultimate fate of submerged oil droplets. To provide a foundation for this work, we have made a series of laboratory measurements using a broadband, high-frequency, calibrated echo sounder. Measurements of oil droplet frequency-dependent target strength were made in a 6 m deep tank of fresh water. Target strength was measured and compared to droplet size and density. Droplet size ranged from 60 μm to 1 mm and was measured by high definition camera. Oils of different density were used including castor, gasoline, diesel, and crude oil. Sound speed of each oil was measured using a Digibar Pro sound velocimeter.

9:05

4aAO3. Laboratory observations of the target strength of non-spherical gas bubbles in water. Thomas C. Weber, Liam Pillsbury, and Scott Loranger (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, tom.weber@unh.edu)

Naturally occurring methane bubbles in the ocean are often observed to have ellipsoidal or otherwise deformed shapes. Models of acoustic scattering from non-spherical gas bubbles suggest changes in resonance frequency and scattering strength relative to spherical gas bubbles with identical volumes. These changes potentially confound our ability to easily relate measurements of acoustic backscatter from gas bubbles to their size and quantity. To help quantify the magnitude of this effect, we have conducted a series of laboratory measurements of acoustic backscatter from non-spherical air bubbles rising from the bottom of a 6 m deep test tank. The acoustic measurements were made at frequencies between 10 kHz to 150 kHz, well above the frequency of bubble resonance (as is often the case for measurements of methane bubbles in shallow coastal environments). Laboratory measurements of bubbles with different sizes and deformations are compared with models for spherical bubbles with identical volumes.

9:20

4aAO4. Low frequency scattering from dynamic fish schools based on collective animal behavior modeling. Simón E. Alfaro, Jorge Cellio (Ingenieria, Universidad Tecnologica de Chile, Av. Vitacura 10.151, Vitacura, Santiago - Chile, Santiago 7650033, Chile, simon_alfaroj@hotmail.com), Maria P. Raveau (Ingenieria Hidraulica y Ambiental, Pontificia Universidad Catolica de Chile, Santiago, Chile), and Christopher Feuillade (Instituto de Fisica, Pontificia Universidad Catolica de Chile, Santiago, Chile)

Low frequency acoustic scattering from swim bladder fish is dominated by the monopole resonance response of the bladder. A school scattering model has previously been developed [Feuillade *et al*., J. Acoust. Soc. Am. **99**(1), 196–208 (1996)] to predict levels of scattering from schools of bladder fish, which includes multiple scattering effects among the fish, and coherent summation of their radiated fields. In order to incorporate these acoustic interactions, the relative locations of the individual fish within the school are required as an input. To provide a realistic description of fluctuating levels of scattering from schools, a self-organizing model of group formation in three-dimensional space has been developed, based on biological principles of collective animal behavior [Couzin *et al*., J. Theor. Biol. **218**, 1–11 (2002)]. In this model, organization within the school is a function of alignment, and repulsive and attractive tendencies based upon the position and orientation of the individual fish. The results of using this model to simulate the fish behavior demonstrate the spatial and temporal dynamics of the fish school, and indicate how these influence the statistical variability of the acoustic scattering response as a function of frequency. [Work supported by ONR.]

9:35

4aAO5. Bayesian environmental inversion of airgun modal dispersion using a single hydrophone in the Chukchi Sea. Graham A. Warner, Stan E. Dosso, Jan Dettmer (School of Earth and Ocean Sci., Univ. of Victoria, 3800 Finnerty Rd., Ste. A405, Victoria, British Columbia V8P 5C2, Canada, sdosso@uvic.ca), and David E. Hannay (JASCO Appl. Sci., Victoria, British Columbia, Canada)

This paper presents estimated water-column and seabed parameters and uncertainties for a shallow-water site in the Chukchi Sea, Alaska, from trans-dimensional Bayesian inversion of the dispersion of water-column acoustic modes. Pulse waveforms were recorded at a single ocean-bottom hydrophone from a small, ship-towed airgun array during a seismic survey. A warping dispersion time-frequency analysis is used to extract relative mode arrival times

4a THU. AM

as a function of frequency for source-receiver ranges of 3 and 4 km, which are inverted for the water sound-speed profile (SSP) and subbottom geoacoustic properties. The SSP is modeled using an unknown number of sound-speed/depth nodes. The subbottom is modeled using an unknown number of homogeneous layers with unknown thickness, sound speed, and density, overlying a halfspace. A reversible-jump Markov-chain Monte Carlo algorithm samples the model parameterization in terms of the number of water-column nodes and subbottom interfaces that can be resolved by the data. The estimated SSP agrees well with a measured profile, and seafloor sound speed is consistent with an independent headwave arrival-time analysis. Environmental properties are required for anthropogenic noise modeling studies in the Chukchi Sea and for improving acoustic localization of marine mammals detected with passive acoustic monitoring systems.

9:50

4aAO6. Measuring the acoustic scattering response of small groups of live fish in a laboratory tank. Maria P. Raveau, Christopher Feuillade (Pontificia Universidad Católica de Chile, Vicuña Mackenna 4860, Macul, Santiago 7820436, Chile, mpraveau@uc.cl), Gabriel Venegas, and Preston S. Wilson (Dept. of Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Assessing the validity of measurement/model comparison in fisheries acoustics is difficult, due to the uncertainty in ground truth for acoustic measurements obtained in the open water. One way to overcome this is to utilize laboratory measurements, where fish school parameters may be more well known. The primary purpose of this work was to investigate the feasibility of measuring the acoustic properties of a small group of live fish in a laboratory tank using a steady state subtraction method [J. Acoust. Soc. Am. **112**, 1366–1376 (2002)]. Transfer function measurements were obtained in a fresh water tank that contained an enclosed group of goldfish (*Carassius auratus auratus*), in order to describe their resonance scattering behavior. The experimental results were compared with an existing predictive model [J. Acoust. Soc. Am. **99**, 196–208 (1996)], which incorporates both multiple scattering effects between fish, and coherent interaction of their individual scattered fields. Computational modeling, experimental details and data/model comparison will be presented. This technique can be extended to larger tanks and other fish species. [Work supported by ONR.]

10:05–10:20 Break

10:20

4aAO7. Application of time-warping to passive acoustic remote sensing. Oleg A. Godin (Physical Sci. Div, CIRES, Univ. of Colorado and NOAA Earth System Res. Lab.,325 Broadway, Mail Code R/PSD99, Boulder, CO 80305-3328, oleg.godin@noaa.gov), Justin S. Ball (CIRES, Univ. of Colorado, Boulder, CO), Michael G. Brown (Rosenstiel School of Marine and Atmospheric Sci., Univ. of Miami, Miami, FL), Nikolay A. Zabotin, Liudmila Y. Zabotina (CIRES, Univ. of Colorado, Boulder, CO), and Xiaoqin Zang (Rosenstiel School of Marine and Atmospheric Sci., Univ. of Miami, Miami, FL)

Interferometry of ambient and shipping noise in the ocean provides a way to estimate physical parameters of the water column and the seafloor without employing any controlled sound sources. With noise interferometry, two-point cross-correlation functions of noise serve as the probing signals and replace the Green's function measured in the active acoustic remote sensing. The amount of the environmental information, which can be obtained with passive remote sensing, and robustness of the estimates of the water column and the seafloor parameters are expected to increase, when contributions of individual normal modes can be resolved in the noise cross-correlation function. Using the data obtained in the 2012 noise-interferometry experiment in the Straits of Florida [M. G. Brown *et al.*, Geophys. Res. Lett. **41**, 5555–5562 (2014)], this paper demonstrates the feasibility of normal mode decomposition of the noise cross-correlation function measured by two hydrophones. The normal modes are resolved by using time-warping, a signal processing technique that has been previously successfully employed to separate normal modes generated by a wide-band compact sound source in shallow-water waveguides. The passively measured dispersion curves of acoustic normal modes are inverted for geoacoustic parameters of the seafloor. [Work supported by NSF and ONR.]

10:35

4aAO8. Spectrum of sound intensity fluctuations due to mode coupling in the presence of moving nonlinear internal waves and bottom parameters estimation. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, Mt. Carmel, Haifa 31905, Israel, bkatsnels@univ.haifa.ac.il), Valery Grigorev (Phys., Voronezh State Univ., Voronezh, Russian Federation), and James Lynch (WHOI, Woods Hole, MA)

Temporal fluctuations of intensity of sound pulses (300±30 Hz) are studied in the presence of nonlinear internal waves (NIW) moving at the some angle with direction of an acoustic track providing mode coupling. In the episode of Shallow Water 2006 experiment, considered in the work, angle between wave front of NIW and source-receiver direction is about 10 degrees. It was shown that there is main maximum in the spectrum measured at fixed depth ~8 cph and set of smaller peaks (~15 cph, ~25 cph, and ~35 cph) in accordance with the theory proposed by authors earlier. Besides, there is maximum in the spectrum of total intensity (summarized over depth or over all hydrophones of vertical line array) at the frequency ~15 cph. In authors opinion, the last one is determined by coupling of propagating modes (excited by the source) having maximal difference in modal attenuation coefficients (in given case modes 2 and 4). Role of horizontal refraction is small. Using the spectrum speed of NIW and bottom attenuation coefficient (in our case ~0.2 dB/wavelength) are determined. Mentioned parameters are in a good agreement with estimations obtained by other methods. [Work was supported by BSF.]

10:50

4aAO9. Direct inversion for sediment sound speed in ocean acoustics. Tao Lin and Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu)

A fast direct approach for estimating sediment sound speed is presented, building on a previously developed technique. The algorithm inverts for the seabed sound speed employing low frequency measurements. The Deift-Trubowitz integral equation is solved by using sound pressure after a Hankel Transform. A modified Born approximation is implemented that improves on previously obtained results, followed by an interpolation method which provides a further improvement. Although the approaches initially rely on assumptions about the sound speed at infinity, we show that the assumptions can be relaxed without affecting the performance. The methods are stable with noise-free data, but problems arise when noise is added to the acoustic field. It is demonstrated that a regularization technique remedies the problem. [Work supported by ONR.]

11:05

4aAO10. Change-point detection for recursive Bayesian geoacoustic inversions. Bien Aik Tan (DSO National Labs., 14, Sci. Park Dr., Singapore, Singapore 118226, Singapore, tbienaik@dso.org.sg), Peter Gerstoft (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA), Caglar Yardim (ElectroSci. Lab., The Ohio State Univ., Columbus, OH), and William S. Hodgkiss (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA)

In order to carry out geoacoustic inversion in low signal-to-noise ratio (SNR) conditions, extended duration observations coupled with source and/or receiver motion may be necessary. As a result, change in the underlying model parameters due to time or space is anticipated. In this paper, an inversion method is proposed for cases when the model parameters change abruptly or slowly. A model parameter change-point detection method is developed to detect the change in the model parameters using the importance samples and corresponding weights that already are available from the recursive Bayesian inversion. If the model parameter change abruptly, a change-point will be detected and the inversion will restart with the pulse measurement after the changepoint. If the model parameters change gradually, the inversion (based on constant model parameters) may proceed until the accumulated model parameter mismatch is significant and triggers the detection of a change-point. These change-point detections form the heuristics for controlling the coherent integration time in recursive Bayesian inversion. The method is demonstrated in simulation with parameters corresponding to the low SNR, 100–900 Hz LFM pulses observed in the Shallow Water 2006 experiment.

Session 4aBA

Biomedical Acoustics: Acoustic Radiation Force in Biomedical Applications II

Mostafa Fatemi, Cochair
Physiology & Biomedical Engineering, Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905

Matthew W. Urban, Cochair
Department of Physiology and Biomedical Engineering, Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905

Invited Papers

9:30

4aBA1. Measurement of cardiovascular tissue stiffness with acoustic radiation force. Gregg Trahey (Biomedical Eng., Duke Univ., 136 Hudson Hall, Box 90281, Durham, NC 27708, gregg.trahey@duke.edu)

We have developed qualitative and quantitative methods of imaging the stiffness of myocardial and vascular tissues using acoustic radiation force. In ongoing animal and clinical studies, these methods have been utilized to differentiate vulnerable from stable vascular plaques, to monitor the formation of radio-frequency- and cryo-ablation lesions, and to assess the impact of ischemia, infarct, pre-load, afterload, and coronary artery pressure on cardiac stiffness. We present alternative pulse-sequencing techniques and image reconstruction methods and discuss their impact on image contrast and resolution. We discuss the relevance of these methods in assessing cardiovascular disease and the technical challenges of introducing them into clinical practice.

9:50

4aBA2. New parameters in shear wave elastography. JeanLuc Gennisson, Thomas Deffieux, Mathieu Pernot, Mathias Fink, and Mickael Tanter (ESPCI, CNRS, INSERM, Institut Langevin, 1 Rue Jussieu, Paris 75005, France, jl.gennisson@espci.fr)

In the field of shear wave elastography, a specific technique called Supersonic Shear Imaging (SSI) was developed since almost 15 years. This technique is based on two concepts: By means of the acoustic radiation pressure phenomena shear waves are generated directly within tissues. Then, shear wave propagation is caught in real time by using an ultrafast ultrasound device (up to 20,000 frames/s). As shear wave speed is directly related to stiffness of tissues, such a concept allows to recover elastic maps of organs. Nevertheless, stiffness is not always only sufficient to better understand organs pathologies and behaviors. So, there is a need to add new parameters for a better characterization. In this context, SSI technique can be extended in order to reach new mechanical parameters, which can potentially help physicians. By looking at the shear wave dispersion, viscosity of tissues can be retrieved by using the right rheological model. Elastic anisotropy is recovered by rotating the probe at the surface of the investigated organ. For each position, the shear wave speed is calculated allowing to deduce orientation of fibers. At last, the change in tissue stiffness as a function of the pressure applied over medium, also called acoustoelasticity theory, allows the assessment of the nonlinear elastic properties. The combination of all these new parameters, viscosity, anisotropy, and nonlinearity, with stiffness offer new possibilities of diagnosis for physicians to better understand organs pathologies.

10:10

4aBA3. Using acoustic radiation force to probe tissue mechanical properties: Challenges and strategies. Stephen A. McAleavey (Biomedical Eng., Univ. of Rochester, 309 Goergen BME/Optics Bldg., Rochester, NY 14627, stephen.mcaleavey@rochester.edu)

The acoustic radiation force phenomenon provides a highly flexible instrument with which to probe the mechanical properties of tissue. The ability to shape the applied force both in time and space, combined with the high sensitivity of ultrasound motion tracking, enables a multitude of techniques for the estimation of tissue mechanical properties. A present challenge is to reduce the measurement variance of these properties so as to better distinguish subtle variations associated with disease stage. Shear wave tracking is the starting point for many of these techniques, and can be used to characterize tissue in terms of group velocity, as well in terms of tissue constitutive model parameters. This talk will discuss sources of error in shear wave velocity and tissue parameter estimation and strategies for their avoidance or compensation. Time-domain estimation of viscoelastic parameters using a wave propagation model and maximum likelihood estimator will be described, along with challenges due to non-ideal source geometries.

10:30

4aBA4. Acoustic radiation force to reposition kidney stones in humans. Michael R. Bailey, Bryan W. Cunitz, Barbrina L. Dunmire (Ctr. Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, bailey@apl.washington.edu), Jonathan D. Harper, Franklin H. Lee, Ryan Hsi, Mathew D. Sorensen (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), James E. Lingeman (Dept. of Urology, Indiana Univ. Health, Indianapolis, IN), Maria M. Karzova, Petr V. Yuldashev, Vera A. Khokhlova, and Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

This is a report of the first clinical study to reposition kidney stones with acoustic radiation force. Studies were conducted with the approval of the University of Washington IRB and a U.S. FDA Investigational Device Exemption. Of the 15 subjects, average age was 56 ± 11 years; average BMI was 29 ± 3; and stone size range was dust to 13 mm. Two patients reported skin discomfort and sensation at depth with a few pushes. Otherwise, there was no pain or adverse effects associated with the treatment. Stones were repositioned in 14 subjects. Stones were repositioned to a new location in all 6 post-lithotripsy patients, while 4 of the 6 passed over 30 stone fragments within a few days of treatment. *De novo* stones and stones as large as 8 mm were repositioned. In four of the 15 subjects, what was noted in clinical imaging as a single, potentially unpassable stone was shown to be several passable stones upon repositioning with ultrasound. Ultrasonic propulsion can safely and without pain reposition kidney stones in humans. [Work supported by NIH NIDDK grants DK043881and DK092197 and National Space Biomedical Research Institute through NASA NCC 9-58.]

Contributed Papers

10:50

4aBA5. Simulation of ultrasound radiation force induced shear wave propagation in viscoelastic media using a mapped Chebyshev pseudospectral method. Bo Qiang (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905), John C. Brigham (Dept. of Civil and Environ. Eng., Univ. of Pittsburgh, Pittsburgh, PA), Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI), James F. Greenleaf, and Matthew W. Urban (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN 55905, Urban.Matthew@mayo.edu)

We implemented a three-dimensional solver for simulating the ultrasound radiation force induced shear wave motion in viscoelastic media using a mapped Chebyshev pseudospectral method. The solver used a body force field that is simulated by the Field II program as the excitation. Then, a velocity-stress formulation was used to calculate the shear wave motion. We used a Voigt model to describe tissue viscoelasticity, and the viscosity is introduced by memory variables and auxiliary differential equations. The top boundary of the domain had a fixed boundary condition to mimic the interaction between an ultrasound probe and the tissue surface. All other sides of the domain were surrounded by perfectly matched layers to minimize the reflections. Time stepping was implemented with the LSODE integrator with variable step sizes. Results show that the waveform of the simulated shear wave is close to experimental observations and equivalent FEM simulations. We used a Fourier-based method to estimate the phase velocities of the shear waves between 100–1000 Hz and a nonlinear regression to estimate the Voigt model parameters. The reconstructions matched within 10% error compared with the true values.

11:05

4aBA6. Compensating for Scholte waves in single track location shearwave elasticity imaging. Jonathan Langdon, Karla Mercado, Diane Dalecki, and Stephen McAleavey (Biomedical Eng., Univ. of Rochester, 303 Goergen Hall, Rochester, NY 14627, jonathan_langdon@urmc.rochester.edu)

The estimation of shearwave velocity in biological tissues using Single Track Location Shearwave Elasticity Imaging (STL-SWEI) depends on the assumption that the ultrasonically observed particle displacements are due to the propagation of shearwaves in an approximately infinite space. When this assumption is violated, erroneous estimates of the shearwave speed may occur leading to image artifacts. One particularly troubling error occurs when slowly propagating Scholte waves are generated at solid-fluid interfaces. These interface waves travel at a slower speed than the shearwaves produced in STL-SWEI. However, the signals produced appear similar to that of shearwaves and cannot be readily distinguished in the typical STL-SWEI imaging sequence. Instead, alternative sequences are needed to identify and correct for these anomalous wave types. In this work, the surface wave phenomena is examined in the context of STL-SWEI imaging. The appearance of these waves is demonstrated in simulation, tissue mimicking phantoms, engineered tissues, and in liver tissue using a clinical scanner. The effect of the ultrasound beam geometry on the Scholte wave measurement is studied. Finally, a revised STL-SWEI reconstruction method utilizing Radon Sums is presented. Using this method, the simultaneous characterization of shear and Scholte waves is demonstrated for the above materials.

11:20

4aBA7. Shear wave propagation and elasticity imaging of soft tissues under compression. Dae Woo Park, Man M. Nguyen, and Kang Kim (Dept. of Medicine, Univ. of Pittsburgh, 567 Scaife Hall 3550 Terrace St., Pittsburgh, PA 15213, parkd2@upmc.edu)

In efforts to improve detection sensitivity of shear wave elasticity imaging of target tissue lesions with relatively small mechanical contrast to the background tissues, shear wave propagation characteristics in tissues under compression loading have been studied. A finite element hyperelastic tissue model was constructed to characterize the changes of propagating shear wave subject to different mechanical loading and to guide *in vitro* experiments. The shear wave speed sharply increased in an inclusion from 2.4 m/s to 6.3 m/s while it increased from 2.0 m/s to 4.0 m/s in the background tissue with overall compression loading from 0% to 30%. Increased shear wave reflection at the boundary of the inclusion due to increased mechanical contrast was lowered using a directional filter. *In vitro* experiments were performed using a soft phantom block (0.5% agar with 5% gelatin) that contains a hard inclusion (1.5% agar with 5% gelatin) of a long cylinder (D: 8 mm). The reconstructed shear modulus of the inclusion exhibited noticeable nonlinearity, in contrast to linear increase of shear modulus in the surrounding phantom. As a result, the elastic modulus contrast of the inclusion to the surrounding phantom was increased from 0.47 to 1.41 at compression from 0% to 30%.

11:35

4aBA8. Plane nonlinear shear waves in relaxing media. John Cormack and Mark F. Hamilton (UT Austin, Appl. Res. Lab., Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, jcormack@utexas.edu)

Due to very low shear moduli for soft tissue or tissue-like media, shear waves propagate very slowly, on the order of meters per second, making it relatively easy to produce shear waves exhibiting waveform distortion and even shock formation. Finite amplitude effects in plane shear waves result from cubic nonlinearity, compared with quadratic nonlinearity in compressional waves. Both attenuation and dispersion also significantly affect propagation of shear waves in tissue. Here we account for these complex viscoelastic effects by considering a medium with one relaxation mechanism. An analytical solution similar to that of Polyakova, Soluyan, and Khokhlov [Sov. Phys. Acoust. **8**, 78 (1962)] for a compressional wave with a step shock in a relaxing medium is obtained for a shear wave with a step shock in a relaxing medium. The wave profile with cubic nonlinearity

closely resembles that with quadratic nonlinearity. For weak nonlinearity the solution reduces to an expression obtained by Crighton [J. Fluid Mech. **173**, 625 (1986)] for a Taylor shock in a viscous medium with cubic nonlinearity. Numerical simulations are presented comparing shock formation with quadratic and cubic nonlinearity for other wave profiles in relaxing media. [Work supported by the ARL:UT McKinney Fellowship in Acoustics.]

11:50

4aBA9. Three-dimensional finite difference models of shear wave propagation in isotropic, homogeneous soft tissue. Yiqun Yang (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI 48824, yiqunyang.nju@gmail.com), Matthew W. Urban (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, Rochester, MN), and Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI)

Shear wave elastography with ultrasound applies an acoustic radiation force to generate shear waves in viscoelastic soft tissue. To address the need for more effective simulation tools that model shear waves generated by an applied acoustic radiation force, three-dimensional (3D) finite difference programs that simulate propagation of shear waves in an isotropic, homogeneous medium have been created. These programs simulate shear wave propagation in elastic and viscoelastic soft tissue models. The 3D finite difference implementation combines an explicit time-stepping approach and a staggered spatial grid with an absorbing boundary condition that reduces boundary reflections. The acoustic radiation force in these simulations is quickly and accurately simulated in FOCUS (http://www.egr.msu.edu/~fultras-web) for a focused linear ultrasound array with an f/# of 2. The compressional wave speed is 1500 m/s and the shear wave speed is 1.5m/s in the elastic and viscoelastic tissue models, and the shear viscosity is 1 Pa·s in the viscoelastic model. The simulation completes 90,000 time steps of a 10 ms simulation in an 80 mm $\times$ 80 mm $\times$ 80 mm region with spatial sampling equal to the wavelength of compressional component in three days on a 3.4 MHz Intel i7 processor. [Supported in part by NIH Grants R01EB012079 and R01DK092255.]

THURSDAY MORNING, 21 MAY 2015 COMMONWEALTH 1, 10:00 A.M. TO 11:05 A.M.

Session 4aEA

Engineering Acoustics and Physical Acoustics: Funding Opportunities with the National Science Foundation

Allan D. Pierce, Chair
PO Box 339, 399 Quaker Meeting House Road, East Sandwich, MA 02537

Chair's Introduction—10:00

Invited Paper

10:05

4aEA1. Sensors, dynamics, and control: Program overview and relevance to acoustics research. Massimo Ruzzene (National Sci. Foundation, 270 Ferst Dr., Atlanta, Georgia 30332, ruzzene@gatech.edu)

The talk will provide an overview of the sensors, dynamics, and control program, its goals, its funding levels, and its priorities. Specifically, the talk will address topics and research directions that are relevant to the acoustics community at large. Such topics include vibration and noise control, dynamic-based structural health monitoring, wave propagation in complex media, nonlinear dynamics, and acoustic metamaterials. Examples of currently funded projects and ideas for future investigations will be provided to facilitate the discussion. The SDC program supports fundamental research on the analysis, measurement, monitoring, and control of dynamic systems, including development of new analytical, computational and experimental tools, and novel applications to engineered and natural systems. Dynamics is the science of systems that change in time. Control concerns the use of external influences to produce desired dynamic behaviors. Diagnostics concerns the use of observation to infer information about a dynamic system. Objectives of the SDC program are the discovery of new phenomena and the investigation of innovative methods and applications in dynamics, control and diagnostics. The intellectual merit of proposals submitted to the SDC program will be evaluated on the basis of fundamental innovation in foundational areas, on alignment with the core disciplines of the CMMI Division, and on potential for transformative impact within and across disciplinary boundaries.

Session 4aED

Education in Acoustics Public Relations, and Student Council: Expanding Acoustics Outreach with Social Media

Andrew A. Piacsek, Cochair
Physics, Central Washington University, 400 E. University Way, Ellensburg, WA 98926

Andrew C. Morrison, Cochair
Natural Science Department, Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431

Chair's Introduction—8:00

Invited Papers

8:05

4aED1. Twitter's not just for teenagers: A scientists' guide to getting started with Twitter in just 5 minutes a day. Laura Kloepper (Dept. of Neurosci., Brown Univ., 185 Meeting St. Box GL-N, Providence, RI 02912, laurakloepper@gmail.com)

Twitter is an effective, powerful tool for scientists looking to keep abreast of trends in their field, communicate their research, and act as public advocates of science. With a simple 140-character Tweet, a person can announce new publications, comment on current events, share photos and updates of research, connect with budding young scientists, and more. Many people think that using Twitter is a massive time investment, but many basic Twitter activities can be conducted in the time it takes you to drink your morning coffee. This talk will introduce the basics of Twitter, show the power of Twitter, and demonstrate best practices for using Twitter with science—all in just 5 minutes a day.

8:25

4aED2. Understanding the social media medium. Andrew T. Pyzdek (Acoust., The Penn State Univ., PO Box 30, State College, PA 16804, atp5120@psu.edu)

With social media platforms like Twitter, Facebook, and Tumblr largely dictating how information is viewed and shared online, Marshall McLuhan's famous statement that "The medium is the message" has never been more true. While it is possible to share the same message over multiple platforms with minimal variation, success is borne when a unique approach is tailored to each one. This talk will focus on the affordances of each social media platform and how those should inform not only your use of social media but also your choice of platform. Special attention will be payed to common uses for social media in acoustics: disseminating original research, establishing connections within the field, advocating science, and educating a general audience on acoustics. Also presented is my personal experience educating a lay-person audience about acoustics as a moderator of the AskScience subreddit and the curator of the Listen To This Noise blog on Tumblr, and how these platforms differ both in content and form.

8:45

4aED3. Using social media tools efficiently and effectively for acoustics outreach and education. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

The ability to quickly and easily reach a large public audience through social media outlets such as Twitter and Facebook makes these platforms ideal for raising awareness of acoustics-related studies and issues. Some researchers may be unable to dedicate a lot of time toward building and maintaining a social media presence. To make finding and disseminating information more efficient on social media sites, there are a series of tools, which can be combined together to automatically aggregate and post information online in ways which can save valuable time and effort. In this presentation, I will show my tool chains that are used to discover interesting acoustics-related news stories for sharing online and how they can be scheduled for posting over several days. I will also discuss surprising ways in which the automatic discovery of acoustics-related questions by non-scientists make for interesting tutorials to be used in acoustics classes at the undergraduate level.

9:05

4aED4. Professional development with social media. Andy Rundquist (Phys., Hamline Univ., 1536 Hewitt Ave., MS B1807, Saint Paul, MN 55104, arundquist@hamline.edu)

I was lonely. I realized that my colleagues didn't always share my particular passions about teaching and research. Reaching out online using twitter, the Global Physics Department, and blogs has helped me find a community that supports me, challenges me, entertains me, and teaches me. Prompts like "Help! I teach normal modes tomorrow and I need better examples" and "how fast do whistlers settle on a frequency" draw me into the community where I hear several, sometimes conflicting, views that then motivate me to learn. My requests about things like whether minimizing the area between a sound wave and a particular sine wave is akin to finding a Fourier coefficient (it isn't) connect me with experts who don't roam my hallways. I'll talk about how I engaged with my various learning communities and how I've developed various workflows to leaverage them.

9:25

4aED5. Using Twitter to disseminate acoustics information: Planning and measurement using marketing science and sociophysics. Lawrence Norris (Arlington Sci. and Technol. Alliance, 6704G Lee Hwy., Arlington, VA 22205, lnorris@arlingtonscience.org)

While many people use Twitter for many different purposes, it is an amazing tool for disseminating scientific information. There are several serious tools for marketing products, information, and people via Twitter. Indeed, a body of science in marketing and sociophysics is available to plan campaigns on Twitter and to measure their efficacy. In this talk, I will discuss tools and methods that the acoustics community can use to engage science students at all levels, their teachers, and parents, as well as other acousticians, policy-makers, and the general public.

9:45

4aED6. @BYUAcoustics: Using social media to enhance research and outreach at BYU. Blaine M. Harker, Tracianne B. Neilsen, Kent L. Gee, Jennifer K. Whiting, Mark L. Berardi, Pauline White, Nicholas D. Ortega, and Matthew F. Calton (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, blaineharker@byu.net)

Social media has become an increasingly common method of attracting the attention of fellow researchers and promoting interest in acoustics worldwide. A concerted effort has been recently made by the Acoustics Research Group at Brigham Young University to promote acoustics with various methods of enhanced communications. Web page articles were developed with lay-language introductions to the group's research activities, which can be shared directly to social media sites. The group's Facebook page and research Twitter account @BYUacoustics provide information about current research meetings, publications, and acoustics in the news, which help network with students who may be interested in joining the acoustics program and which keep alumni informed of current events. In addition, our outreach efforts have been augmented by use of social media. A YouTube channel has been created that contains videos of acoustical demonstration from our outreach show. A general audience Twitter account @Sounds2Astound has been utilized to connect with students in our descriptive class and to K-12 teachers who often bring their students to tour our facilities. The overall effectiveness of each system is assessed using webpage statistics, analytics, and perceived success in reaching target audiences. Successes and limitations are summarized and lessons learned are outlined.

10:05–10:50 Break

10:50

4aED7. The broader impact of practicing communication through social media: From Twitter to the National Science Foundation. Alexis B. Rudd (Alexis Rudd, Univ. of Hawaii at Manoa, 47-420 Hui Iwa St. #B-304, Kaneohe, HI 96158, rudd@hawaii.edu)

Available research funding has decreased in line with the Budget Control Act of 2011, which resulted in funding cuts across both defense and non-defense discretionary programs. These cuts have had a negative impact on many researchers, including those at the Acoustical Society of America. Legislation and appropriations by congress have a direct effect on the funding levels and research priorities of federal agencies. Clear and relatable communication of these impacts is important to both the public and the members of the US Congressional committees, most of whom do not have a background in scientific research. Consideration of the technical and educational background of the audience is vital to clear communication, and agencies such as the National Science Foundation (NSF) are placing increasing emphasis on the broader impacts of scientific proposals and how research will benefit the people of the United States. Social media is an opportunity for scientists to get real-time feedback on science communication and to practice translating scientific jargon for an audience of non-specialists and explaining technical concepts succinctly (often in 140 characters or less).

11:10

4aED8. Reddit Science AMA: Using "the front page of the internet" to share your knowledge with the world! Laura Kloepper (Dept. of Neurosci., Brown Univ., 185 Meeting St. Box GL-N, Providence, RI 02912, laurakloepper@gmail.com) and Andrew Pyzdek (Penn State Univ., University Park, PA)

The social media service and link aggregator reddit is one of the most popular and influential websites on the internet. Boasting 174 million unique visitors per day, a high percentage of those high school and college aged students, reddit provides the perfect audience for science outreach. One of the features of reddit is Science AMA, or Ask Me Anything, in which scientists answer questions about their work. Participation in these Q&As allows scientists to communicate with an international audience of all ages and backgrounds, and is an effective tool for scientific outreach with limited time investment. In this talk, we will share the outcome of a Bioacoustics AMA conducted at the Indianapolis meeting, and learn how more individuals and TCs can participate in future Science AMAs.

11:30–12:00 Panel Discussion

Session 4aNS

Noise, ASA Committee on Standards, Psychological and Physiological Acoustics, Physical Acoustics, and Structural Acoustics and Vibration: Wind Turbine Noise I

Paul D. Schomer, Cochair
Schomer and Associates Inc., 2117 Robert Drive, Champaign, IL 61821

Nancy S. Timmerman, Cochair
Nancy S. Timmerman, P.E., 25 Upton Street, Boston, MA 02118

Kenneth Kaliski, Cochair
RSG Inc, 55 Railroad Row, White River Junction, VT 05001

Robert D. Hellweg, Cochair
Hellweg Acoustics

Chair's Introduction—8:30

Invited Papers

8:35

4aNS1. Some pitfalls to be avoided in a wind turbine noise research program. Paul D. Schomer (Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

The Acoustical Society of America has created a public policy position relative to the acoustic emissions from wind turbines. This position calls for research that definitively will show if problems exist, and if so, who is affected, how are they affected, and why. Much of the research to date is based on assumptions, frequently contrary to fact or unproven. That is not the kind of research that the ASA desires. The money spent on this questionable research should have been directed towards definitive research such as that envisioned by ASA. This paper talks about some of the previous research and elucidates on their assumptions with the purpose of preventing mistaken test designs like these in the future, and with the purpose of improving the research program to be developed by ASA.

8:55

4aNS2. Health and well-being related to wind turbine noise exposure: Summary of results. David Michaud (Health Canada, Canadian Federal Government, 775 Brookfield Rd., Ottawa, ON J9H7J9, Canada, david.michaud@hc-sc.gc.ca)

This paper summarizes the results of the Health Canada Wind Turbine Noise and Health Study. One participant between the ages of 18–79 years was randomly selected from each household. The final sample included 1238 participates (606 males) living between 0.25 and 11.22 km from wind turbines. The response rate was 78.9% and did not significantly vary across sampling strata or between provinces. Wind turbine noise (WTN) exposure was not found to be related to hair cortisol concentrations, resting blood pressure, resting heart rate, or any of the measured sleep parameters. Self-reported results obtained through an in-person questionnaire do not provide support for an association between increasing WTN levels and self-reported sleep disturbance, use of sleep medication, or diagnosed sleep disorders. Similarly, no significant association was found between WTN levels and self-reported migraines, tinnitus, dizziness, diabetes, asthma, hypertension, perceived stress, or any measure of quality of life. Statistically significant exposure-response relationships were observed between increasing WTN levels and an increase in the prevalence of long term high annoyance towards several wind turbine features, including: noise, shadow-flicker, visual impacts, and vibrations. The influence of background noise on annoyance and the association between WTN annoyance and other reported and measured outcomes is presented.

9:15

4aNS3. Indoor and outdoor narrowband measurements of low frequency and infrasonic noise from wind turbines. Allan Beaudry and Michael Bahtiarian (Noise Control Eng., Inc., 799 Middlesex Turnpike, Billerica, MA 01821, allanb@noise-control.com)

The acoustic impact from industrial-grade wind turbines was studied for turbines located in Cape Cod, Massachusetts. Narrowband and one-third-octave band infrasonic and low frequency measurements were performed at a residence located in proximity to the Vestas V82-1.65 MW wind turbines. Surveys were undertaken inside and outside the residence on multiple occasions under various wind speeds and wind directions. In each case, measurements were performed using an infrasonic microphone prior to, during, and following the nightly shutdown period of the turbines. The spectral results show the existence of discernible tones occurring at the blade passage frequency and its harmonics during operation and absent with the turbines secured. Additional tones where found in the very low audible frequency range with sidebands separated by the blade passage frequency. Comparisons have been made of the indoor and outdoor measurements as well as the effect of wind speed and direction on tone amplitude and prominence. The use of overall G-weighted noise levels to characterize these low frequency tones is also examined.

9:35

4aNS4. Changes in ambient sound levels observed in relation to the operational status of the Mesa Wind Project Site. Jessica Briggs (Fish, Wildlife, and Conservation Biology, Colorado State Univ., Fort Collins, CO), Dr. Megan F. McKenna, Kurt M. Fristrup, (Natural Sounds and Night Skies Div., National Park Service, 1201 Oakridge Dr., Ste. 100, Fort Collins, CA 80525, kurt_fristrup@nps.gov)

Type 1 sound level measurements and continuous audio recordings were obtained from two sites near the Mesa Wind Project Site (MWPS), 100 m and 1500 m from the nearest turbine. The sites were upwind of the turbines. Unexpectedly, MWPS ceased operating 29 days after this project began; data were collected for an additional 127 days. Contrasts in hourly summaries of 1 second 1/3rd octave sound level measurements, controlled for hourly mean wind speed, revealed the contributions of rotating blades. For the near site, at wind speeds below 5 m/s, 1/3rd octave band levels increased starting at about 100 Hz, with distinct spectral peaks appearing between 400 Hz and 2000 Hz. These spectral peaks were less prominent when hourly mean wind speed was 5 m/s or greater. At the far site, small increases in sound level were observed for 1/3rd octave bands at 1 kHz and below when the blades were turning. The levels measured in this study will be compared with levels measured at NPS sites, to evaluate distances at which operational wind farms might affect park acoustical environments.

9:55

4aNS5. Do wind turbines cause adverse heat effects? A review of the evidence. Robert Y. McMurtry (Surgery, Western Univ., 403 Main St., Picton, OntarioN K0K2T0, Canada, rymcmurtry1@gmail.com)

There is a continuing debate about the veracity of AHE in the environs of WT. Proponents claim there is insufficient evidence to support the existence of direct AHE and therefore WT are safe to erect near human habitation. The international regulations governing WT placement are highly variable but generally a setback of at least 350 m (WHO 500m) and a noise limit of 40 dBA are recommended. Nonetheless there is mounting evidence that AHE are in fact occurring. In the 1980s, N D Kelley produced evidence of direct causation of AHE, especially sleep disturbance, from downwind WT, which was attributable to impulsive infrasound and low frequency noise. Subsequently, there has emerged further evidence regarding upwind WT that satisfies the nine Bradford Hill criteria. The presentation will define AHE and address each Braford Hill criterion.

10:15–10:25 Break

10:25

4aNS6. "Waking in an anxious frightened panicked state" and "pressure bolt sensations"—what clinical and acoustic clues can tell us about the causes of deteriorating physical and mental health. Sarah E. Laurie (Waubra Foundation, PO Box 7112, Banyule, Melbourne, Victoria 3084, Australia, sarah@waubrafoundation.org.au)

Residents living in quiet rural areas report a variety of new symptoms correlating with exposure to new acoustic emissions from industrial sound and vibration sources, including large wind turbines, of which sleep disturbance is the most common. Some residents and their health practitioners report a unique pattern of sleep disturbance, which indicates physiological stress is occurring concurrently. Other clinical and acoustic clues such as "pressure bolt" sensations correlating with independently measured pressure pulses will also be shared. Neurophysiological pathways which appear to be activated by these acoustic stimuli will be discussed, followed by reference to the extensive body of clinical research supporting the current knowledge that both chronic sleep deprivation and chronic stress directly exacerbate pre existing diseases, as well as directly cause new diseases resulting in serious and sometimes irreversible damage to physical and mental health. Finally, specific research to investigate these physiological and pathological impacts, in order to determine safe threshold exposure levels for long term exposure to impulsive infrasound and low frequency noise will be proposed.

10:45

4aNS7. Why are regulators, communities, neighbors, and acousticians annoyed by wind turbines? Stephen E. Ambrose (SE Ambrose & Assoc., Windham, ME) and Robert W. Rand (Rand Acoust., 1085 Tantra Park Circle, Boulder, CO 80305, rrand@randacoustics.com)

Noise assessments for large wind turbines should be more reliable based on decades of human-response research. Empirical data clearly show the most severe noise impacts occur in the quieter environments. Populations living in quiet areas were the least researched until the arrival of wind turbines. European studies reveal that the public has a greater sensitivity to wind-turbine noise than transportation sounds. Acoustic researchers developed the percent highly annoyed metric to model. This metric should be shaped by an assessment of the acoustic environment using many variables all subjective, debatable, and mathematical. A review of dose-response research will attempt to lay the groundwork for a new noise assessment methodology specific to wind turbines.

11:05

4aNS8. Understanding the human impact caused by the sound of wind turbines. William K. Palmer (TRI-LEA-EM, 76 SideRd. 33-34 Saugeen, RR 5, Paisley, Ontario N0G2N0, Canada, trileaem@bmts.com)

A mystery surrounds the human impact reported by those with wind turbines in their environment. “PubMed” identifies 23 papers for “wind turbine human health.” A dose-response relationship between exposure to sound, annoyance, and sleep disturbance is generally accepted. Self-reporting identified other impacts that commenced or increased with wind turbines operation. Recent articles repeat that no peer-reviewed papers show other links than annoyance and possibly sleep disturbance, and suggest that wind turbine visibility, negative attitudes, fear, or lobby groups cause adverse reporting, but provide no evidence dispelling impacts. Many argue acceptability of annoyance is a social measure set by government. Meantime, the Government of Canada “Wind Turbine Noise and Health Study” presented findings of an association between increasing noise from wind turbines and annoyance, but no found no evidence linking exposure to wind turbine noise to any of the self-reported illnesses, and no association between wind turbine noise and measures of stress, sleep quality, or significant changes in quality of life. Neglected in all of this, those who have been adversely impacted believe no one is listening. This paper takes up their position, to examine other acoustic factors to generate a hypothesis for cause of the human impact.

Contributed Paper

11:25

4aNS9. Annoyance of wind-turbine noise as a function of amplitude-modulation parameters. Christina Ioannidou, Sébastien Santurette, and Cheol-Ho Jeong (Dept. of Elec. Eng., Tech. Univ. of Denmark, Ørsteds Plads, DTU Bygning 352, Kgs. Lyngby 2800, Denmark, ses@elektro.dtu.dk)

Amplitude modulation (AM) has been suggested as an important factor for the perceived annoyance of wind-turbine noise (WTN). Two AM types, typically referred to as “normal AM” and “other AM,” depending on the AM extent and frequency region, have been proposed to characterize WTN AM. The extent to which AM depth, frequency, and type affect WTN annoyance is a matter of debate. In most subjective studies, the temporal variations of WTN AM have not been considered. Here, a sinusoidally modulated WTN model accounting for temporal AM variations was used to generate realistic artificial stimuli in which the AM depth, frequency, and type, while determined from real on-site recordings, could be varied systematically. Subjective listening tests with such stimuli showed that a reduction in AM depth, quantified by the modulation depth spectrum, led to a significant decrease in annoyance. When the spectrotemporal characteristics of the original far-field stimuli were included in the model and the temporal AM variations were taken into account by varying the modulation index over time, neither AM frequency nor AM type were found to significantly affect annoyance. These findings suggest that the effect of AM parameters on WTN annoyance may depend on the intermittent nature of WTN AM.

THURSDAY MORNING, 21 MAY 2015 KINGS 1, 8:30 A.M. TO 12:05 P.M.

Session 4aPA

Physical Acoustics: Infrasound I

Roger M. Waxler, Chair

NCPA, University of Mississippi, 1 Coliseum Dr., University, MS 38677

Invited Papers

8:30

4aPA1. Incorporating atmospheric uncertainties into estimates of the detection capability of the IMS infrasound network. Alexis Le Pichon (CEA, DAM, DIF, Arpajon F-91297, France, alexis.le-pichon@cea.fr)

To monitor compliance with the Comprehensive Nuclear-Test-Ban Treaty (CTBT), a dedicated network is being deployed. Multi-year observations recorded by the International Monitoring System (IMS) infrasound network confirm that its detection capability is highly variable in space and time. Today, numerical modeling techniques provide a basis to better understand the role of different factors describing the source and the atmosphere that influence propagation predictions. Previous studies estimated the radiated source energy from remote observations using frequency dependent attenuation relation and state-of-the-art specifications of the stratospheric wind. In order to account for a realistic description of the dynamic structure of the atmosphere, model predictions are further enhanced by wind and temperature error distributions as measured in the framework of the ARISE project. In the context of the future verification of the CTBT, these predictions quantify uncertainties in the spatial and temporal variabilities of the IMS infrasound network performance in higher resolution, and will be helpful for the design and prioritizing maintenance of any arbitrary infrasound monitoring network.

8:50

4aPA2. Infrasound study of the atmospheric fine-scale wind velocity structure and its variability. Igor Chunchuzov, Sergey Kulichkov, Vitaly Perepelkin, Oleg Popov (Obukhov Inst. of Atmospheric Phys., 3 Pyzhevskii Per., Moscow 119017, Russian Federation, igor.chunchuzov@gmail.com), Jelle Assink (2CEA/DIF/DAM/DASE in Bruyères-le-Châtel, Arpajon, France), and Roger Waxler (3National Ctr. for Physical Acoust., University, MS)

The results of study of the fine-scale wind velocity structure in the upper stratosphere, mesosphere, and lower thermosphere by using recently developed method of infrasound probing of the atmosphere are presented. The method is based on the effect of infrasound scattering from highly anisotropic wind velocity and temperature nonhomogeneities in the middle and upper atmosphere. The vertical profiles of the wind velocity fluctuations in the upper atmosphere (up to a height of 140 km) are retrieved from the wave forms and travel times of the infrasound signals from volcanoes and surface explosions. The vertical wavenumber spectra of the retrieved wind velocity fluctuations are obtained for the upper stratosphere. Despite the difference in the location of explosive sources all the obtained spectra show the existence of high vertical wavenumber spectral tail with certain power law decay. The effect of a fine-scale wind velocity structure and its variability on the wave forms, coherence, and frequency spectra of the infrasound arrivals is studied. The possibility to use retrieved wind velocity structure in the upper stratosphere and lower thermosphere for improving an infrasound monitoring of infrasound sources in the atmosphere and parameterizing statistical characteristics of anisotropic turbulence (variances, spatial and temporal spectra, and coherence) is discussed.

9:10

4aPA3. Absorption of infrasound and acoustic-gravity waves in the atmosphere. Oleg A. Godin (Physical Sci. Div., CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., 325 Broadway, Mail Code R/PSD99, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

Long-range propagation of infrasound and acoustic-gravity wave fields in the middle and upper atmosphere are strongly affected by air viscosity and thermal conductivity. To characterize the wave dissipation, it is typical to consider idealized environments, which admit plane-wave solutions. This paper presents an asymptotic approach that relies instead on the assumption that spatial variations of environmental parameters are gradual. Unlike the traditional approach, the asymptotic theory allows one to derive AGW dispersion equations in a consistent manner for a wide range of scenarios and to describe the wave attenuation more realistically. Realistic assumptions about the atmosphere are found to lead to rather different predictions for absorption of atmospheric waves than the plane-wave solutions. Wind speed and wave frequency appear only through the intrinsic wave frequency in the dispersion equation, including the terms that describe the effects of viscosity and thermal conductivity. It is found that the Sutherland-Bass model of infrasound absorption should be modified to account for wind-induced absorption anisotropy, which arises from deviations of the intrinsic frequency from the wave frequency that is observed in a stationary reference frame. The anisotropy is expected to result in a significant decrease in the predicted attenuation of thermospheric returns.

9:30

4aPA4. Acoustic multipole source inversions of volcano infrasound. Keehoon Kim, David Fee (Geophysical Inst., Univ. of Alaska Fairbanks, 903 Koyukuk Dr. (Rm. 307A), Fairbanks, AK 99775, keehoon.kim@gmail.com), Jonathan M. Lees (Univ. of North Carolina Chapel Hill, Chapel Hill, Alaska), Akihiko Yokoo (Kyoto Univ., Kumamoto, Japan), and Mario C. Ruiz (Instituto Geofísico de la Escuela Politécnica Nacional, Quito, Ecuador)

Sources of volcano infrasound involve the atmospheric displacement associated with volcanic eruptions, where characteristic source dimensions are generally confined by the vent. Volcano infrasound sources are typically considered as a monopole which corresponds to the first-order term in the acoustic multipole expansion. However, when the wavelength becomes comparable to the size of the vent, the source may have further complexity which can be described only by higher-order terms, yet such complexity of volcano infrasound source has not been extensively explored. This is mainly due to (1) limited sampling of the acoustic wavefields due to poor network coverage and (2) complex sound propagation near the volcanic edifice which significantly distorts the multipole acoustic wavefields. In this study, we present a linearized waveform inversion technique incorporating numerical Green's functions. A full 3-D Finite-Difference Time-Domain (FDTD) method accelerated with GPU is used to compute accurate Green's functions taking into account volcano topography. The presented method is applied to infrasound data recorded at Sakurajima volcano (Japan) and Tungurahua volcano (Ecuador) and volcano infrasound sources associated with explosive eruptions are characterized in terms of a monopole and dipole. These methods could be applied to chemical explosions as well to determine source characteristics and complexity.

9:50

4aPA5. Spall effects on infrasound generation from explosions at the Nevada National Security Site. Kyle R. Jones (Sandia National Labs., 1515 Eubank Ave. SE, Albuquerque, NM 87123, krjones@sandia.gov), Arthur J. Rodgers (Lawrence Livermore National Lab., Livermore, CA), Rodney W. Whitaker (Los Alamos National Lab., Los Alamos, NM), Souheil M. Ezzedine, and Oleg Y. Vorobiev (Lawrence Livermore National Lab., Livermore, CA)

We apply two methods to evaluating the spall signature from underground chemical explosions such as those at the Source Physics Experiment (SPE) at the Nevada National Security Site (NNSS). The first approach uses the Rayleigh integral to compute overpressures for buried explosions from synthetic vertical acceleration data at surface ground zero. To obtain the acceleration data, we systematically vary parameters such as the spall duration, depth of burial and magnitude and observe the effect on the resulting acoustic waveform shape. The second method uses a hydrodynamic approach to more fully characterize the varied parameters to produce the acoustic waveforms. As the spall decreases we find that the acoustic waveform shape changes dramatically. This waveform signature may provide diagnostics on the explosive source and may be a useful metric for underground explosion monitoring. [This work was done under award number DE-AC52-06NA25946. Sandia National Laboratories is a multi-program laboratory managed and operated by Sandia Corporation, a wholly owned subsidiary of Lockheed Martin Corporation, for the U.S. Department of Energy's National Nuclear Security Administration under contract DE-AC04-94AL85000.]

10:10

4aPA6. Infrasound from underwater sources. Läslo Evers (KNMI / TU Delft, PO Box 201, De Bilt 3730 AE, Netherlands, evers@knmi.nl), David Brown (CTBTO, Vienna, Austria), Kevin Heaney (OASIS, Lexington, MA), Jelle Assink (CEA, DAM, DIF, Arpajon, France), Pieter Smets (KNMI / TU Delft, De Bilt, Netherlands), and Mirjam Snellen (TU Delft, Delft, Netherlands)

Atmospheric low-frequency sound, i.e., infrasound, from underwater events has not been considered thus far, due to the high impedance contrast of the water-air interface making it almost fully reflective. Here, we report for the first time on atmospheric infrasound from a large underwater earthquake (Mw 8.1) near the Macquarie Ridge, which was recorded at 1325 km from the epicenter. Seismic waves coupled to hydroacoustic waves at the ocean floor, after which the energy entered the Sound Fixing and Ranging channel and was detected on a hydrophone array. The energy was diffracted by a seamount and an oceanic ridge, which acted as a secondary source, into the water column followed by coupling into the atmosphere. The latter results from evanescent wave coupling and the attendant anomalous transparency of the sea surface for very low frequency acoustic waves.

10:30–10:45 Break

10:45

4aPA7. Probabilistic infrasound propagation using ensemble based atmospheric perturbations. Pieter Smets and Läslo Evers (Seismology and Acoust., Royal Dutch Meteorological Inst., PO Box 201, De Bilt 3730 AE, Netherlands, smets@knmi.nl)

The state of the atmosphere is of utmost importance for infrasound propagation. In propagation modeling, the true state of the atmosphere is mainly represented by the analysis. The analysis is the best deterministic estimate of the atmosphere using a data assimilation system existing of a General Circulation Model (GCM). However, the analysis excludes error variances of both model and observations. In addition, the coarse resolution of GCM results in averaging of, e.g., clouds or gravity waves, over larger regions known as parameterization. Consequentially, arrivals due to fine-scale structure in wind and temperature can be missing. Therefore, infrasound propagation including the atmospheric best-estimate error variances based on an ensemble model is proposed. The ensemble system exists of model perturbations with an amplitude comparable to the analysis error estimates to obtain a probability density function. The best-estimate analysis error variances are described by a set of perturbations using the European Centre for Medium-range Weather Forecasts (ECMWF) Ensemble Data Assimilation (EDA) system. Probabilistic infrasound propagation is demonstrated by one year of mining activity, e.g., blasting, in Gällivare, northern Sweden, observed at infrasound array IS37 in Norway, part of the International Monitoring System (IMS) for verification of the Comprehensive Nuclear-Test-Ban Treaty (CTBT). Probabilistic infrasound propagation is compared with the standard deterministic result obtained using the analysis.

Contributed Papers

11:05

4aPA8. Infrasonic analysis of the October 28, 2014 Antares rocket failure at Wallops Island, Virginia, using video recordings as ground truth. Jay J. Pulli and Aaron Kofford (Raytheon BBN Technologies, 1300 North 17th St., Ste. 400, Arlington, VA 22209, jpulli@bbn.com)

We used close-in video recordings of the October 28, 2014 Antares rocket failure at Wallops Island, VA to establish an event timeline to aid in the analysis of infrasound recordings made at nearby stations of the IRIS Transportable Array. Our timeline is ignition at 22:22:38 UTC, liftoff at +4 s, bright plume and first explosion at an altitude of 300 m at +15 s, second large explosion as the rocket hits the ground at +25 s, followed by the excess fuel burn lasting some 400 s. Both explosions and the fuel burning events are seen in the infrasound data recorded at IRIS station S61A at a distance of 23 km, and the two explosions can be seen out to distances of 130 km. High resolution time frequency analyses of the infrasound signals at the distant stations show a dispersed signal from 0.5–8 Hz with a peak at 1.7 Hz and corresponding group velocity of 360 m/s. This dispersion curve corresponds to a low velocity duct at the surface with a thickness of approximately 1.2 km. The relatively fast group velocity can be attributed to the prevailing winds. Explosion yield estimates using the BOOM model indicate equivalent TNT yields of 20 and 200 tons for the two explosions. Co-located seismic and infrasound sensors at two stations allow us to estimate the acoustic-to-seismic spectral ratio at 1–10 μm/s/Pa. However, low coherence between the acoustic and seismic signals implies a non-linear transfer function at the sites.

11:20

4aPA9. Infrasound propagation and model reduction in randomly layered media. Christophe MILLET (CEA, DAM, DIF, Arpajon 91297, France, christophe.millet@cea.fr)

A consensus has emerged within the infrasound research community that gravity waves are filtered out in the available atmospheric models. Apart from occasionally strong lower-atmospheric effects, the major wave influences occur in the middle atmosphere, between 10 and 110 km altitudes. In the present approach, the unresolved gravity waves are represented as a random field that is superimposed on the average background state, and the wave equation is solved using a reduced-order model, starting from the classical normal mode technique. The reduced model is obtained by retaining a few propagating modes, with the aim of simplifying the acoustic model to the point that the predicted statistics of waveforms are correct, even though some small irrelevant details is lost. We focus on the asymptotic behavior of the transmitted waves in the weakly heterogeneous regime, for which the coupling between the wave and the medium is weak. The statistics of a transmitted broadband pulse are computed by decomposing the original pulse into a sum of modal pulses that can be described by a front pulse stabilization theory. Specifically, it is shown how reduced-order models can be used to explain some aspects of the variability of large infrasound datasets.

11:35

4aPA10. Sound induced plume instability. Konstantin A. Naugolnykh (Phys., Univ. of Colorado, 325 Broadway, Boulder, CO 80305, konstantin.naugolnykh@noaa.gov)

A sustained source of buoyancy creates a continuous rise of lighter fluid through the ambient denser fluid, with mixing occurring along the way. Such structure, called as a plume, is sensitive with respect to flow perturbations. In particular, the effect of sound can modulate the structure of plume as a result of sound-turbulent interaction. The acoustic wave can slightly change the structure of flow and then interact with in-phase spatially modulated turbulence. In application to plume this effect is considered in the present paper follow the approach of Chimonas, 1972 and Moiseev *et al*., 2000.

11:50

4aPA11. Hamiltonian ray tracing in an arbitrary curvilinear coordinate system—Amplitude estimation. Philip Blom and Stephen J. Arrowsmith (EES, Los Alamos National Lab., Los Alamos National Lab., PO Box 1663, Los Alamos, NM 87545, pblom@lanl.gov)

Acoustic ray tracing is known to be an efficient and robust numerical method to compute propagation paths of acoustic energy through the atmosphere and ocean. Over large propagation distances, a Cartesian formulation of ray tracing is known to be inadequate and a spherical or spheroidal formulation is required in order to obtain accurate propagation paths. An overview of the ray tracing equations in an arbitrary curvilinear coordinate system will be presented and compared with the known spherical coordinate formulation. Typically, coaxial ray paths are used to compute ray tube density and provide an estimation of the amplitude along the ray path. Here, the method of auxiliary parameters is used to identify the exact geometric spreading factor along individual ray paths. The implementation of the resulting ray tracing methods will be discussed including methods used to decrease computation time and planned additions to the propagation scheme.

THURSDAY MORNING, 21 MAY 2015 KINGS 4, 8:00 A.M. TO 12:45 P.M.

Session 4aPP

Psychological and Physiological Acoustics and Speech Communication: Influence of Visual Cues on Auditory Perception

Jonas Braasch, Chair
School of Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

Chair's Introduction—8:00

Invited Papers

8:05

4aPP1. Understanding cross-modal interactions of spatial and temporal information from a cortical perspective. Barbara Shinn-Cunningham, Samantha Michalka, Abigail Noyce, and David Somers (Boston Univ., 677 Beacon St., Boston, MA 02215-3201, shinn@bu.edu)

In the ventriloquism effect, perception of visual spatial information biases judgments of auditory events, yet there is little effect of auditory stimuli on perception of visual locations. In the flash-beep illusion, the number of auditory "beeps" biases judgments of the count of visual flashes, but visual flashes have little effect on perceived number of auditory events. These asymmetries suggest that vision is the "natural" sensory modality for coding spatial information, while audition is specialized for representing temporal information. Here, we review recent behavioral and neuroimaging evidence from our labs exploring the neural underpinnings of such perceptual asymmetries. Specifically, we find that there are distinct frontal cortical networks associated with visual information and auditory information. Yet these networks can be recruited by the other sensory modality, depending on task demands. For instance, when judging spatial aspects of auditory inputs, neural areas associated with visual processing are recruited; when judging temporal aspects of visual inputs, areas associated with auditory processing are activated. We also find another asymmetry: knowing *when* a spatial event is going to occur helps listeners judge location, but knowing *where* an event will occur does not help judgments about that event's timing. These kinds of studies help elucidate how temporal and spatial information is encoded in the brain, and the neural mechanisms by which visual-spatial and auditory-temporal information interact.

8:25

4aPP2. Localizing sound sources when the listener moves: Vision required. William Yost and Xuan Zhong (Speech and Hearing Sci., ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

In an article under review at JASA, we showed that when the listener rotates, determining the rotational direction of sound moving from loudspeaker to loudspeaker or if the sound is fixed at one loudspeaker location requires two forms of information: information provided the auditory spatial cues (e.g., interaural differences) AND information about the position of the head/body. Primary head/body position cues are provided by vision (and to some extent by the vestibular system and probably by other sensory and cognitive systems). In this paper, we address a very simple question which, as far as we know, has not been asked before: What is sound source localization accuracy when the listener moves? A sound source localization identification task was used to measure root-mean-square-errors (rms) in degrees in the full azimuth plane. Different stimuli were used and listeners were either rotated at a constant velocity or were stationary. They listened with their eyes open or closed. When listeners had no visual or others forms of information about head/body position (e.g., eyes closed) and, thus, all they had were the auditory spatial cues (e.g., interaural difference) sound source localization accuracy was very poor. [Research supported by an AFOSR grant.]

8:45

4aPP3. Visual capture of a stereo image. Yi Zhou and Christopher Montagne (Speech and Hearing Sci., Arizona State Univ., 975 S. Myrtle Ave., Coor 3470, Tempe, AZ 85287, yizhou@asu.edu)

This study investigated visual bias in localizing a "phantom" sound source generated by a stereo pair of speakers. The lateral position of the fused auditory image was controlled by varying the time delay or intensity ratio of two 15-ms noise bursts presented from two hidden loudspeakers positioned at +/− 45 degrees in azimuth. Visual stimuli were delivered from three LEDs positioned at −45, 0, and 45 degrees in azimuth. A multiple-alternative, forced choice task was implemented to measure listener's perceived sound localization with and without LED stimulation. The magnitude of visual capture (bias) was compared for the two stimulus manipulations—time delay and intensity ratio. Eleven normal hearing listeners participated in the task. Results showed stronger visual capture for time delay than intensity ratio manipulation. Binaural signal analyses were further conducted on ear-canal signals recorded from a KEMAR, showing that time delay manipulation introduced ambiguous binaural and spectral cues. The degree of visual bias was found to be correlated with the ambiguity of localization cues in the task.

9:05

4aPP4. Auditory/visual distance perception. Pavel Zahorik (Heuser Hearing Inst. and Div. of Communicative Disord., Dept. of Surgery, Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu)

A considerable body of research suggests that the perception of sound source distance exhibits systematic biases. The distances of far sources are progressively underestimated, but near sources are overestimated. Such biases are typically not observed in vision, where perceived distance is found to be highly accurate under natural viewing conditions in which a variety of visual distance cues are available. Relatively little is known about how distance information from both auditory and visual modalities is combined in the perception of distance, however. This is surprising, given that auditory/visual aspects of directional perception have been extensively studied, primarily in relation to the "ventriloquist effect." Here, two recent experiments on auditory/visual distance perception are summarized. The results from both suggest that not only is perceived distance less accurate in the auditory modality than in vision, but it is also considerably less precise. These results explain why visual information, when available, appears to dominate auditory information in the perception of distance. [Work supported by NEI.]

9:25

4aPP5. The interconnection of binaural and stereoscopic cues for distance perception. Kristen Murphy (Rensselaer Polytechnic Inst., 124 Fulton St. East, 2nd Fl., Grand Rapids, MI 49503, kmurphy@acousticsbydesign.com)

Encountering objects within our three-dimensional environment is a combination of sensory cues. While much research has been conducted on how the auditory and visual systems singularly locate objects in space, studies in cross-modal interaction of these two systems suggest the overall perception of a stimulus cannot fully be described by a single sensory system. The purpose of this study was to examine the relationship between visual and acoustic cues specifically in understanding depth perception. Previous studies of audio-visual interaction of spatial perception have accounted for many different auditory depth cues, but have not fully cross examined a number of visual depth cues. Participants were asked to determine the egocentric distance of a stimulus in a virtual three dimensional environment in terms of real-size quantitative scale. Stimuli were presented as either audio only, visual only, or audio and visual. The single-modal and multi-modal results were analyzed and compared against the null hypothesis that when presented simultaneously, the visual depth cues dominate the distance judgment and the acoustic depth cues have no bearing. In this study, the null hypothesis was not rejected, in accord with previous research investigating the proximity image effect.

9:45

4aPP6. Non-phonemic benefits of visual cues in the perception of consonants. Douglas Brungart, Lynn M. Bielski (Walter Reed NMMC, 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.brungart@us.army.mil), and Eric R. Thompson (Air Force Res. Lab., Dayton, OH)

Visual cues from the talker's face improve speech perception because the talker adopts discrete facial configurations, known as visemes, corresponding to a limited number of possible phonemes in the auditory signal. Visual cues alone are insufficient for complete recovery of the speech signal, because individual visemes can occur with more than one phoneme. Visual speech cues may provide non-phonemic benefits in noise through segregation of target speech from background sounds. This experiment isolated phonemic and non-phonemic benefits of visual cues through identification of strings of consonants (i.e., aCHaBaGa) in an eight-alternative forced choice task. Listeners discriminated each consonant presented from a randomly selected foil in each position. In the heterovisemic case, the foil had a different viseme than the target. In the homovisemic case, each foil was from a list of consonants with the same viseme as the target, meaning the visual target provided no phonetic information about the speech. Results show benefit from visual speech cues in the homovisemic condition roughly half the size of that in the heterovisemic condition. This suggests a substantial portion of the visual benefit can be attributed to factors other than extraction of phonemic information from visemes in the visual stimulus.

10:05–10:15 Break

10:15

4aPP7. Individual differences in real-time processing of audiovisual speech by preschool children. Tina M. Grieco-Calub (The Roxelyn and Richard Pepper Dept. of Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., FS, 2-231, Evanston, IL 60208, tinagc@northwestern.edu) and Janet Olson (School of Allied Health and Communicative Disord., Northern Illinois Univ., DeKalb, IL)

Visual speech cues maximize speech perception for adults in challenging listening environments: reliance on visual speech increases with greater auditory degradation. Young children, however, may be limited in audiovisual speech perception because of the cognitive costs associated with multimodal processing. In this study, real-time speech understanding was measured in preschool children ($N = 33$, 30–48 months of age) using eye-tracking methodology in the absence (quiet) or presence of two-talker babble and in the absence (auditory-alone, AO) or presence of audiovisual (AV) speech cues. On each trial, children were instructed by a female speaker to look at one of two objects projected onto a large screen. Speech processing was quantified by how quickly children fixated the target object (reaction time, RT) and overall accuracy of target-object fixation. Visual benefit was calculated as the difference in performance between the AO and AV conditions. Analyses revealed negative correlations between RT_{AO} and RT_{AV} in both the quiet and two-talker babble conditions: visual speech facilitated speech processing in children with slow RT_{AO}, but not in children with fast RT_{AO}. Results suggest that visual speech facilitates speech perception in preschool children, but is likely dependent on children's processing efficiency of the AO speech signal.

10:35

4aPP8. Auditory scene analysis with a multi-modal model. Thomas Walther (Institut für Kommunikationsakustik, Ruhr-Universität Bochum, Universitätsstraße 150, Bochum 44801, Germany, thomas.walther@rub.de)

By integrating state-of-the-art auditory scene-analysis methods with artificial cognitive feedback, the current project challenges human performance in quality of experience tasks and auditory scene analysis. Human listeners are regarded as multi-modal agents that develop their concept of the world by exploratory interaction. The prominent goal of the project is to develop an intelligent, computational model of active auditory perception and experience in a multi-modal context. The resulting system framework will form a structural link from binaural perception to judgment and action, realized by interleaved signal-driven (bottom-up) and hypothesis-driven (top-down) feedback processing within an innovative expert-system architecture. A conceptual overview of the project framework is presented, and insight is given into the current state of research, focusing on CASA-related search-&-rescue (S&R) scenarios. In these scenarios, an autonomous robot is endowed with auditory/visual feature-analysis facilities that provide it with bottom-up information of its environment. Top-down evaluation of the extracted features then results in cognitive feedback loops that allow the machine to adapt to complex S&R scenarios and perform significantly better than would be possible by only employing feed-forward control mechanisms. [Work performed in the context of the EU project Two!Ears.]

10:55

4aPP9. Considering, capturing, and representing the visual environment in noise perception studies. J. Parkman Carter (Architectural Acoust., Rensselaer Polytechnic Inst., 32204 Waters View Circle, Cohoes, NY 12047, cartej8@rpi.edu)

Despite studies which show the profound influence of visual environments on aural expectations and perceived noise annoyance, popular noise mapping techniques continue to reflect only one dimensional sound pressure level metrics. Contextual factors, which significantly influence noise perception, can be gleaned from the visual environment, but there are no established protocols for documenting and representing salient visual features. Techniques for capturing and representing the visual environment will be discussed, which can address the intrinsic differences between audio and photographic capture in terms of dynamic range and spatial resolution. Immersive audio and video projection techniques which are currently being developed at RPI's Collaborative Research Augmented Immersive Virtual Environment (CRAIVE) Lab will also be presented. These large scale multi-modal presentation strategies significantly improve efforts to conduct subject studies in environmental noise perception and assessment.

11:15

4aPP10. Visual influence on the subjective impressions of urban soundscapes. Tyler Adams (Architectural Acoust., Rensselaer Polytechnic Inst., 1402 N Mariposa Ave. #8, Los Angeles, CA 90027, echotyler@gmail.com)

Audio recording is a common method for evaluating subjective impressions of soundscapes, which is useful because the same information can be reproduced and presented to subjects in a controlled environment. This study was conducted to determine what impacts visual information might have in the subjective evaluations of urban soundscapes. Audio of urban environments was recorded using binaural and multi-channel methods; video was simultaneously recorded using a single channel digital camera. The soundscapes were reproduced for subjects in a laboratory environment with and without the accompanying videos. The data gathered from evaluations made in the laboratory were also compared with evaluations conducted *in situ* to determine the degree to which impressions might change in a controlled environment.

11:35

4aPP11. Defying the physical: Acoustic design as a medium for reimagining space and recontextualizing expression. Bobby E. Gibbs (Dept. of Commun. Sci. and Disord., Univ. of South Carolina, Columbia, SC 29208, artfull.mind@gmail.com) and Jonas Braasch (School of Architecture, Commun. Acoust. and Aural Architecture Lab, Rensselaer Polytechnic Inst., Troy, NY)

Just as the physical laws of light are both an inspiration and a point of departure for visual artists, the canvas of acoustic design is not aesthetically bound to real spaces. Using findings from venue visits, interviews and an interactive virtual auditory exploration, we will discuss the bricolage of physical space and cultural expression inherent in the dissemination of a unique musical genre. In particular, we will explore how experimental improvisers preserve and subvert spatial cues to create aural experiences that are at once intimate and illusive.

11:55

4aPP12. A bi-modal model to simulate auditory expectation for reverberation time and direct-to-reverberant energy from visual feedback. Jonas Braasch, M. T. Pastore, Nikhil Deshpande (Ctr. for Cognition, Commun., and Culture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu), and Jens Blauert (Inst. of Commun. Acoust., Ruhr Univ. Bochum, Bochum, North Rhine-Westphalia, Germany)

A bi-modal model is presented that predicts the psychophysical results of Valente and Braasch [Acustica, 2008]. The model simulates expectation of room-acoustical qualities due to visual cues. The visual part of the model estimates angles of incidence and delays of the first two side reflections of a given frontal sound source. To this end, a stereoscopic image is used to determine azimuth angles and distances for the two frontal room corners. The distance estimates are derived by using the angular differences between the left- and right-eye images of each corner. The model then calculates the room volume by reconstructing a rectangular room from these data, assuming a range of possibilities for the missing room coordinates. In a next step, logarithmic fits of volume to expected reverberation time and of volume to direct-to-reverberant energy ratio predict the expected value ranges for these two parameters. Using a feedback structure, the visually-derived acoustic parameters become input to an auditory Precedence-Effect model, where they are used to pre-adjust inhibition parameters for two acoustic side reflections. These inhibition parameters are consequently refined in the course of the analysis of the incoming sound. [Work supported by NSF #1229391/#1320059 and ERC FP7-ICT-2013-C-#618075.]

Contributed Papers

12:15

4aPP13. Auditory illusions arising from lack of visual clues. Steven J. Waller (Rock Art Acoust., 5415 Lake Murray Blvd. #8, La Mesa, CA 91942, wallersj@yahoo.com)

Auditory illusions arising from lack of visual clues will be discussed and related to prehistoric cave art and Stonehenge. In the ancient past, when the wave characteristics of sound were not understood, virtual sound effects arising from complex sound wave interactions (echoes, reverberations, interference patterns, etc.) were misinterpreted as invisible beings (echo spirits, thunder gods, sound absorbing bodies, etc.) as described in ancient myths around the world. Scientifically conducted experiments involving blindfolded participants show how various ambiguous sounds can be interpreted in more than one way—like optical illusions. The blindfold in this case is a metaphor for ignorance of the wave nature of sound. It is proposed that ancient peoples were looking at the apparent source of sound and not seeing anything, and so concluded that the sounds emanated from beings that could not be seen. These experiments thus can help in understanding our ancestors' perceptions and reactions to sounds they considered mysterious and spooky. These discoveries are just a few examples of research findings that are springing from the new field of Archaeoacoustics. See https://sites.google.com/site/rockartacoustics/ for further examples.

12:30

4aPP14. Dynamic binaural sound source localization with ITD cues: Human listeners. Xuan Zhong, William Yost (Speech and Hearing Sci., Arizona State Univ., 975 S Myrtle Ave., Lattie F Coor Hall 2211, Tempe, AZ 85287, xuan.zhong@asu.edu), and Liang Sun (Dept. of Elec. and Comput. Eng., The Univ. of Texas at San Antonio, San Antonio, TX)

In real life, human listeners rarely experience cone of confusion errors in localization of sound sources due to the limitation of interaural time difference (ITD) as a spatial hearing cue. Previous work on robotics suggested that the confusion is disambiguated as successive observation of ITD is made over time, when self-motion of the microphone system was allowed. In this behavioral study, we investigated whether horizontal/vertical planes localization with time difference is possible when self-motion is allowed for human listeners. In particular, the case of a static sound source playing low frequency pure tone signal was studied. Human listeners were seated in a rotating chair in the middle of a loudspeaker array, and a low frequency audio signal was played at an elevation. The task was to identify the elevation spatial angle under conditions when vision was allowed and denied. The hypothesis was, with vision, a much more robust observation of self-motion is available, and the report of elevation would be more accurate. The results largely agreed with the hypothesis. A similar study for robotic hearing generated similar results.

Session 4aSA

Structural Acoustics and Vibration and Noise: Noise Identification and Control in the Mining Industry

Marcel Remillieux, Cochair
Los Alamos National Laboratory, Geophysics Group (EES-17), Mail Stop: D446, Los Alamos, NM 87545

Hugo E. Camargo, Cochair
Office of Mine Safety and Health Res., NIOSH, 626 Cochrans Mill Road, Building 155, Pittsburgh, PA 15236

Invited Papers

8:00

4aSA1. Noise identification, modeling, and control in mining industry. Debi Prasad Tripathy (Dept. of Mining Eng., National Inst. of Technol.,Rourkela, Odisha 769008, India, dptripathy@nitrkl.ac.in) and SANTOSH KUMAR NANDA (Ctr. for Res., Development and Consultancy (CRDC), Eastern Acad. of Sci. and Technol., Bhubaneswar, India)

Prolonged exposure of miners to the high levels of noise in opencast and underground mines can cause noise induced hearing loss and non-auditory health effects. To minimize noise risk, it is imperative to identify machinery noise and their impacts on miners at the work place and adopt cost effective and appropriate noise control measures at the source, path, and at the receiver. In this paper, authors have summarized the noise levels generated from different machineries used in opencast and underground mines and elaborated on frequency dependent noise prediction models, e.g., ISO 9613-2, ENM, CONCAWE, and non-frequency based noise prediction model VDI-2714 used in mining and allied industries. The authors illustrated the applications of innovative soft computing models, viz., Fuzzy Inference System [Mamdani and Takagi Sugeno Kang (T-S-K)], MLP (multi-layer perceptron, RBF (radial basis function) and adaptive network-based fuzzy inference systems (ANFIS) for predicting machinery noise in two opencast mines. The paper highlights the developments and research conducted on effective noise control measures being adopted for mining machineries and implemented in mines to minimize the noise menace so that noise levels generated in mines are within the prescribed noise standards and rules.

8:30

4aSA2. Comparison of noise reduction values for fit tests and work in coal mines. Brandon C. Takacs (Safety & Health Extension, West Virginia Univ., 3604 Collins Ferry Rd., Morgantown, WV 26506, brandon.takacs@mail.wvu.edu), Steven E. Guffey (Industrial Management Systems Eng., West Virginia Univ., Morgantown, WV), Mingyu Wu (Occupational Safety and Health Management, Grand Valley State Univ., Grand Rapids, MI), and Kevin Michael (Michael and Assoc., State College, PA)

At present exposure limits, one in four workers will develop a permanent hearing loss as a result of mining coal (Prince 1997). Mine Safety and Health Administration (MSHA) inspectors found in the period of 1986-1992 that approximately 25% of coal miners' daily noise doses exceeded MSHA's PEL. Virtually all mines have hearing conservation programs and virtually all miners are issued and told to wear either ear muffs or ear plugs. Nevertheless, miners still have a high rate of noise induced hearing loss (NIHL). An important question is to what degree the ineffectiveness of hearing conservation programs is due to failure of miners to wear muffs and plugs properly when they are needed and how much is due to inadequacies of hearing protectors. If the former is important, can technological innovations provide means to improve use of muffs and plugs. If the latter is important, can individual fit-testing improve noise reduction (NR) values achieved by miners. A related issue is whether fit-testing in an office environment adequately predicts NR values achieved during work if non-wearing times are excluded. To address those issues, WVU is conducting studies in a lab and in coal mines that primarily involve measuring sound levels in the ear (SPLear) concurrently with sound levels at the shoulder (SPLsh), allowing computation of NR values for protectors.

9:00

4aSA3. Calibration and assessment of noise monitoring instrumentation used by the Mine Safety and Health Administration. John P. Homer (Dept. of Labor, Mine Safety and Health Administration, 626 Cochrans Mill Rd., Pittsburgh, PA 15236, homer.john@dol.gov)

The use of noise dosimeters is the standard method for noise exposure monitoring and assessment in the United States mining industry. The Mine Safety and Health Administration (MSHA) calibrates and maintains a fleet of various noise dosimeters and acoustical calibrators for the purpose of conducting enforcement activities and providing technical support to mine operators. The dosimeter fleet is comprised of both corded- and badge-type instruments. MSHA studies have conclusively revealed evidence in support of both types as to performing acceptably both in practice and when tested against standardized criteria, ANSI S1.25-1991. MSHA operates an acoustical calibration laboratory which maintains National Institute of Standards and Technology (NIST) traceable calibration records for all noise dosimeters and calibrators used for enforcement and support functions. The laboratory is currently in the process of modernization. Replacement test systems will provide increased reliability, reduced turn-around time, and improved accuracy through the

implementation of modern computer-based components. Improvements provided by the laboratory modernization project will greatly improve quality assurance and provide the ability to pursue accredited compliance with ISO/IEC 17025:2005, through the American Association for Laboratory Accreditation (A2LA).

9:30

4aSA4. Vibration exposure characteristics and health risk prevention strategies associated with vibration exposure during mining applications. Tammy R. Eger (Human Kinetics, Laurentian Univ., 935 Ramsey Lake Rd., Sudbury, Ontario P3E 2C6, Canada, teger@laurentian.ca) and James P. Dickey (Kinesiology, Western Univ., London, Ontario, Canada)

Occupational exposure to vibration can lead to health problems and is typically classified as whole-body vibration (WBV), hand-arm vibration (HAV), or foot-transmitted vibration (FTV). Vibration exposure characteristics are typically measured and compared to standards (ISO 2631-1; ISO 5349-1; EU Directive 2002/44/EC) in an effort to determine the probability of adverse health effects including: low-back pain, spinal degeneration, and gastrointestinal tract problems linked with exposure to WBV; decreased grip strength, tingling/numbness in the fingers/hands, and blanching of the fingers associated with exposure to HAV; and numbness/tingling in the feet/toes, and cold induced blanching of the toes stemming from exposure to FTV. The presentation will summarize vibration exposure data collected by the research team over a 10-year period which suggests operators of load-haul-dump vehicles, haulage trucks, and dozers are exposed to WBV above recommended guidelines, while operators of jack-legs are exposed to HAV above guidelines. Our data also suggest miners exposed to FTV associated with drilling off of raise platforms and operating bolters and jumbo drills are at risk of developing a vibration-induced injury. The presentation will also review the effectiveness of control strategies evaluated by the research team including seating, isolation platforms, “anti-vibration” drills, cab interventions, road maintenance, and operating speeds.

10:00–10:15 Break

10:15

4aSA5. Measurement and analysis of noise and acoustic emission on a roof bolter for identification of joints and in rock. Jamal Rostami (Energy and Mineral Eng., The Penn State Univ., 126 Hosler Bldg., University Park, PA 16802, rostami@psu.edu), Soheil Bahrampour, Asok Ray (Pennsylvania State Univ., University Park, PA), and Craig Collins (R&D, JH Fletcher, Huntington, WV)

Interpretation of the data obtained from roof bolt drilling can offer a reliable source of information that can be used to characterize the ground for use in ground support design and evaluation. Drilling for installation of roof bolts is done by various drills and often it is a noisy process. The acoustic emission from the drill can be used as a source for identification of the rock formations and voids/joints in the rock mass. This paper offers a brief review of the ground support using roof bolts, followed by introduction and discussion of the roof characterization methods by instrumented roofbolters. In particular, installation of noise measuring sensors on the drill head to provide additional source of information will be discussed. The analysis and digital filtering of measured noises was used to identify the voids or joints in the full scale testing. A brief overview of the instrumentation and full scale testing, as well as review of the results of initial testing on an instrumented drills will be offered in the paper. The paper will introduce an algorithm for digital filtering of the acoustic emission to identify the discontinuities in the rock mass.

10:45

4aSA6. Field evaluations of a noise control for roof bolting machines. Amanda Azman (NIOSH, 626 Cochrans Mill Rd., Pittsburgh, PA 15226, aazman@cdc.gov)

Roof bolting machine (RBM) noise is a significant health hazard in underground coal mining because the sound levels at the operator’s station often exceed 100 dB(A). Drilling the hole prior to installing a roof bolt is the most significant source of RBM operator noise exposure. The dominant source of drilling noise is the drill steel. NIOSH has developed an enclosure to surround the drill steel as it is drills the hole to reduce the noise reaching the machine operator. The collapsible drill steel enclosure, or CDSE, is designed to collapse upon itself as the drill steel is further advanced into the mine roof so the drill steel can remain enclosed during the entire hole-drilling process. Original versions of this device effectively reduced noise exposure, but were deemed unacceptable by the mining community due to various usability and durability issues. A modified version has been developed and NIOSH laboratory tests revealed a 2–4 dB(A) noise reduction. This paper describes the in-mine evaluation of the redesigned CDSE. The field tests focused on the noise reduction as well as miner acceptance of the device. The results confirm improved usability as well as a 42% reduction in noise dose at the operator position.

11:15

4aSA7. Jackhammer chisel damper. Henry A. Scarton and Kyle R. Wilt (Mech., Aerospace, and Nuclear Eng., Rensselaer Polytechnic Inst., Rensselaer Polytechnic Inst., 110 Eight St., JEC 4008, Troy, NY 12180, scarton@rpi.edu)

This paper will present the performance of a rugged Coulomb Friction damper for a Jackhammer chisel. The predominant source of high intensity airborne noise from a pneumatically muffled jackhammer is the ringing resulting from impacting the undamped chisel, which radiates airborne sound from the impacts of the hammer exciting the transverse bending modes. A simulated steel chisel moil point was constructed with geometric properties similar to a jackhammer chisel and designed so as to survive the severe acceleration impacts from the reciprocating hammer. Anechoic tests of the chisel and it damped equivalent indicate that the sound pressure level for the undamped chisel due to longitudinal impact was 86.8 dB linear (re to 20 Pa) with the strongest ring tone at 1.6 kHz and harmonics; the sound pressure level for the damped chisel with identical axial impacts was reduced by 13 dB to 73.5 dB with severe reduction of chisel ring.

Session 4aSC

Speech Communication: Cross-language Speech Production and Perception (Poster Session)

Rachel M. Theodore, Chair
University of Connecticut, 850 Bolton Road, Unit #1085, Storrs, CT 06269

Posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to view other posters, contributors of odd-numbered papers will be at their posters from 8:15 a.m. to 9:45 a.m., and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 11:30 a.m. There will be a 15-min break from 9:45 a.m. to 10:00 a.m.

Contributed Papers

4aSC1. The role of articulatory cues in the establishment of perceptual categories. Emily Cibelli (Linguist, UC Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, ecibelli@berkeley.edu)

This study investigates the interaction of perceptual and articulatory information in the acquisition of non-native phoneme categories. There is evidence that perceptual learning can improve production of target sounds in a new language, but the reverse—improvement of perceptual discrimination from articulatory learning—is less well-studied (for one example, see Catford and Pisoni 1970). In this experiment, native English speakers learned Hindi coronal stop contrasts (dental vs. retroflex; 4-way VOT contrast) in a pre-test/training/post-test paradigm. Training involved production training, where subjects got explicit instruction about place of articulation and voicing of the target sounds, and practiced producing them. Comparison of pre- and post-test performance on an AX discrimination task indicated greater accuracy (β = 0.486, t = 2.763) and faster responses (β = −0.215, t = −3.396) for most target contrasts after training. (Those contrasts which did not improve tended to map onto English phonemic distinctions, and were well-discriminated at pre-test.) This result supports the hypothesis that articulatory information can contribute to the early development of novel perceptual categories under certain conditions. More generally, it suggests that information from one speech domain can be used to support representations in another speech domain.

4aSC2. Top-down linguistic categories dominate over bottom-up acoustics in lexical tone processing. Tian C. Zhao and Patricia K. Kuhl (Inst. for Learning & Brain Sci., Univ. of Washington, Portage Bay Bldg., Seattle, WA 98195, zhaotc@uw.edu)

Native tonal-language speakers exhibit reduced sensitivity to lexical tone differences within categories when compared to across categories (i.e., a top-down linguistic influence). However, increasing evidence suggests enhanced sensitivities to lexical tones among musically trained individuals (bottom-up influence), though previous studies examined non-tonal language speakers processing lexical tones. We investigated the relative contribution of top-down and bottom-up processing of lexical tones when both strategies are available. Seventeen native Mandarin speakers with extensive musical training completed a music test and a lexical tone discrimination task. The music test validated their enhanced pitch-processing abilities. The lexical tone discrimination task measured participants' lexical tone sensitivities along a continuum. Results were compared to existing data from Mandarin non-musicians and English speaking musicians (Zhao & Kuhl, 2012). Despite their enhanced bottom-up pitch processing, Mandarin musicians' performance exhibited patterns similar to Mandarin non-musicians, reflecting the dominant influence of linguistic categories ($p = 0.504$). Their sensitivities were also lower than English-speaking musicians ($p = 0.011$). However, further regression analysis suggests that bottom-up processing still plays a role in Mandarin musicians: their within-category tone discrimination is predicted by their musical pitch discrimination scores, similar to English musicians but opposite Mandarin non-musicians.

4aSC3. Mandarin Chinese vowel and tone identification in noise: Effects of native English experience. Mingshuang Li, Wenjing Wang, Sha Tao, Qi Dong (State Key Lab. of Cognit. Neurosci. and Learning, Beijing Normal Univ., Rm. 9417, Jingshi Hotel, No.19, Xinjiekouwai St., Haidian District, Beijing 100875, China, limingshuang@mail.bnu.edu.cn), Jingjing Guan, and Chang Liu (Dept. of Commun. Sci. and Disord., Univ. of Texas at Austin, Austin, TX)

Several studies found differences in English vowel identification between Chinese-native listeners in the United States (CNU) and China (CNC) in multi-talkers babble (MTB) and temporally modulated (TM) noise, but not in quiet condition. Two possible explanations were proposed: first, CNU listeners used temporal modulation of noise more efficiently than CNC listeners; second, CNU listeners had less informational masking of MTB than their CNC peers. The current study aimed at exploring whether the difference in noise processing between CNU and CNC listeners was also presented for their native speech perception. Chinese vowel and tone identifications were measured for CNU and CNC in quiet, stationary and TM noise, babble-modulated noise and MTB. The identification scores of CNU listeners were significantly higher than CNC listeners in most noisy backgrounds, whereas both groups had the same performance in quiet. Moreover, compared with CNC listeners, CNU listeners gained greater masking releases from temporal modulation in noise at low SNRs, while the informational masking was comparable between the two groups. In conclusion, the 1–3 year residency in English-speaking country may improve Chinese-native listeners' capacity to use temporal cues in noise, but may not improve their ability against informational masking of English babble when listening to their native speech.

4aSC4. Cross-linguistic vowel variation in Saterland: Saterland Frisian, low German, and high German. Wilbert Heeringa, Heike Schoormann, and Jörg Peters (German Studies, Univ. of Oldenburg, P.O. Box 2503, Oldenburg 26111, Germany, wilbert.heeringa@uni-oldenburg.de)

This study investigates the vowel space of trilingual speakers of Saterland Frisian, Low German, and High German. The three vowel systems show differences in the number of distinct categories but share the majority of vowel qualities. To examine whether the size of the vowel space correlates positively with the number of vowel categories and whether distinctive vowels are positioned in the vowel space so as to increase perceptual contrast (Liljencrants and Lindblom, 1972, Lindblom, 1986) speakers were instructed to read vowels of all three languages in a /hVt/ frame. Measurements of mid-vowel F1 and F2 of monophthongs neither revealed a positive correlation between the size of the vowel space and the inventory size nor cross-linguistic differences of dispersion, except for a higher dispersion of High German vowels in the F2 dimension. Single vowel categories of Saterland Frisian and Low German were merged with respect to formant frequencies and duration. High German showed longer vowel duration and higher F2 of the front vowels than Saterland Frisian and Low German and a lower F1 than Low German. These results suggest that the trilingual speakers use the same phonetic categories for Saterland Frisian and Low German but not for High German.

4aSC5. Articulatory and acoustic correlates of English front vowel productions by native Japanese speakers. Sonya Mehta and William F. Katz (The Univ. of Texas at Dallas, 3531 Hilltop Ln., Plano, TX 75023, naya@utdallas.edu)

Speakers of languages having a relatively small vowel inventory (e.g., Japanese) may experience interference effects when attempting to produce novel vowels from a language with a larger vowel inventory, such as English. Studies of English vowels produced by native Japanese speakers have shown formant frequency and duration differences between native and accented productions. Less is known about the exact articulatory processes underlying second language vowel learning. The current study examines tongue position during native Japanese speakers' productions of American English vowels. Ten speakers produced the front vowels /i/, /ɪ/, /e/, /ɛ/, and /æ/ while seated in a 3D electromagnetic articulograph (EMA) system that tracked the position of the tongue (tongue tip, TT, tongue dorsum, TD, and tongue back, TB) and lower lip during vowel productions in the consonant environment /hVd/. Kinematic and acoustic measures were taken at the midpoint of each vowel steady-state portion. Differences in tongue position for tense-lax vowel pairs were determined by calculating the Euclidean distance between vowel centroids. Preliminary results suggest more spatial overlap in vowels produced by Japanese talkers than those produced by monolingual English speakers. The relationship between these articulatory distances and acoustic measures (formant frequency and durational changes) will be explored.

4aSC6. Mapping vowels from American English to Taiwanese Mandarin: A perceptual study. Yu-sheng Wang, Yueh-chin Chang, and Feng-fan Hsieh (Graduate Inst. of Linguist, National Tsing Hua Univ., No. 101, Section 2, Kuang-Fu Rd., Hsinchu 30013, Taiwan, sam3300086@gmail.com)

It has been noted that vowel backness is largely preserved in English-to-Mandarin loanwords, but not vowel height. This asymmetry contradicts the native phonological patterns. The aim of this study is to re-examine this issue from an experimental perspective. Ten Taiwanese Mandarin (TM) speakers (aged 23–30) participated in this experiment. English Stimuli consisted in /CVmi/ sequences with C corresponding to one of the three consonants {b, d, g} and with V to one of the four vowels {ɛ, ɔ, æ, ɚ}. Mandarin stimuli also consisted in /CVmi/ sequences with C corresponding to one of the three consonants {p, t, k} and with V to one of the following vowels {i, a, u, ei, ie, au, ou, uo}. The participants were asked to rate similarity between American English and TM vowels (on scale 1–7). Our results partially support the generalizations in loanword adaptation, namely that TM speakers tend to map English [æ] onto Mandarin [ei], rather than [a], while [ɔ] is mapped to [ou/uo], [ɛ] to [ei], and [ɚ] to [ɤ]. We further measured the perceptual distance between English and Mandarin vowels by means of Euclidean distance. Given the assumption that smaller acoustic distance means greater perceptual similarity, it turned out that only this pair (English [ɚ] and Mandarin [ɤ]) can be explained away. It is thus concluded that loanword adaptation is not entirely based on raw acoustic signals. Phonological features and phonotactics also play a significant role.

4aSC7. On the fricative vowels in Suzhou Chinese. Fang Hu (Inst. of Linguist, Chinese Acad. of Social Sci., 5 Jian Guo Men Nei St., Beijing 100732, China, hufang@cass.org.cn) and Feng Ling (Shanghai Univ., Shanghai, China)

Apical vowels are widely distributed in Chinese dialects, whereas fricative vowels or strident vowels are less known. This paper is an acoustic and articulatory study of fricative vowels in the Suzhou dialect of Wu Chinese. The acoustic data from 20 speakers show that the fricative vowels have formant patterns between their plain and apical counterparts. Linguographic data from four speakers reveal that more laminal part of the tongue is involved in the production of the fricative vowels than their plain counterparts, which are basically anterodorsal. And the EMA study from three speakers confirms a comparatively advanced lingual configuration in the production of the fricative vowels. Although the production of fricative vowels is characterized by visible turbulent frication from the spectrograms, and a significantly lower harmonics-to-noise ratio, the results suggest that spectral characteristics of fricative vowels and apical vowels play a more important role in defining the vowel contrasts. That is, plain high vowels, fricative high vowels, and apical vowels distinguish in place of articulation, namely, being anterodorsal, laminal, and apical, respectively. And frication could be treated as a concomitant and redundant feature in the production of fricative or apical vowels.

4aSC8. Are opaque patterns what we think they are? The acoustics of the Bengali vowel chain shift. Traci C. Nagle and Kelly H. Berkson (Linguist, Indiana Univ., Memorial Hall 322, Bloomington, IN 47405, tcnagle@indiana.edu)

The chain-shift pattern of vowel height harmony in Bengali, in which low and mid vowels in monosyllabic verb stems alternate with mid and high vowels, respectively (æ~e, e~i, ɔ~o, o~u), is reported in the phonological literature and by Bengali linguists to be exceptionless. Experimental evidence, however, indicates that Bengali speakers extend this pattern to nonce verbs only about half the time when low vowels are involved. This raises questions about the productivity of the attested pattern, echoing the results of experimental tests of chain shifts in other languages (e.g., Polish, Taiwanese) in which speakers fail to extend opaque phonological patterns to nonce words. One question we may ask is whether the phenomenon described in the literature as an alternation truly neutralizes the vowel height contrasts in the Bengali language. This research presents instrumental acoustic analysis of Bengali vowels produced by native speakers in real and nonce verbs in order to address this question and gain new insight into both the quality of the vowels produced as a result of harmony in real words and how speakers apply the pattern with nonce verbs.

4aSC9. Temporal organization of frication in fricativized vowels. Matthew Faytak (Linguist, Univ. of California, Berkeley, 2632 San Pablo Ave., Apt. A, Berkeley, CA 94702, mf@berkeley.edu)

Fricative vowels are vocoids that exhibit a clear formant structure as well as fricative noise produced by a labiodental or (post)alveolar constriction (Connell 2007). *Apical vowels*, similar vocoids found in many Chinese languages, have known (post)alveolar constrictions, but it has been debated whether or not these actually generate substantial fricative noise (cf. Lee-Kim 2014). In order to better understand the relationship between these two segment types, acoustic data was obtained for the fricativized vowels in two Chinese languages (Standard Mandarin and Suzhou Wu) as well as Kom, a Grassfields Bantu language spoken in Cameroon. Analysis of fricativized vowel spectra and timecourses suggests two predominant types, one with steady-state fricative noise of a relatively low intensity, and a second that is more internally dynamic, exhibiting high-intensity frication toward the beginning of the vowel and lower-intensity frication towards the end. Standard Mandarin's apical vowels are shown to be broadly consistent with the second type, with less frication than typically described impressionistically. However, Suzhou Wu and Kom speakers vary between the two types. This suggests that the language-specific phonetic implementation of this fricative noise does not neatly align with any commonly used descriptive terms.

4aSC10. Phonation type contrasts and vowel quality in Marathi. Kelly Berkson (Dept. of Linguistics, Indiana Univ., 1021 E. Third St., Mem 322E, Bloomington, IN 47405, kberkson@indiana.edu) and Stephen Politzer-Ahles (New York Univ., Abu Dhabi, Abu Dhabi, United Arab Emirates)

Phonation-type contrasts in consonants and vowels are typically associated with a number of acoustic differences. Phonemic breathy phonation, for instance, is fairly consistently associated with lower Cepstral Peak Prominence values than modal sounds, and with higher values for spectral measures such as H1-H2 and H1-A3. The effect of breathy phonation on vowel quality, however, is less consistent: lower first formant (F1) values have been found for breathy vowels in some languages, while others do not show consistent F1 differences based on phonation type. Furthermore, at present little work has investigated the effect of breathy voiced consonants on formant values in subsequent vowels. The current study presents data from Marathi, an Indic language with phonemically breathy-voiced obstruents and sonorants. Ten native speakers (five males and five females) produced real words that included the vowels [a] and [e] after both modal and breathy consonants. The effect of consonant phonation type on F1 and F2 values in subsequent vowels is reported in order to augment our understanding of the relationship between voice quality differences and vowel quality.

4aSC11. Persian speakers of English: Acoustics of vowel epenthesis. Christina C. Akbari (Commun. Disord., Arkansas State Univ., PO Box 910, State University, AR 72467, cakbari@astate.edu), Katsura Aoyama (Commun. Disord., Univ. of North Texas, Denton, TX), and James Dembowski (Speech, Lang., and Hearing Sci., Texas Tech Health Sci., Lubbock, TX)

This study results from the need to develop further understanding into the process of vowel epenthesis which is often observed with second language learners. The purpose of this study was to examine the epenthetic vowels produced by Persian speakers of English to determine the acoustical characteristics and to ascertain if these vowels were acoustically different or similar to the main vowels. Past research indicated that at times the vowels could be copies of the main vowels but at other times they were different. However, no acoustical data were provided in regards to the acoustical characteristics of the epenthetic vowels (durations, F1, and F2 frequencies). Twenty Persian speakers of English took part in the study. All participants arrived in the United States after the age of 22. All of the participants demonstrated limited English proficiency on a standardized measure. The results indicated that the epenthetic vowels produced were qualitatively different from the main vowels in terms of duration, F1, and F2 frequencies.

4aSC12. Vowel context effects on the spectral dynamics of English and Japanese sibilant fricatives. Patrick F. Reidy and Mary E. Beckman (Dept. of Linguist, The Ohio State Univ., 24A Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, reidy.16@osu.edu)

Previous analyses of vowel context effects on the sibilant fricatives of English (/s, ʃ/) and Japanese (/s, ɕ/) have focused on spectral properties computed from a limited number of time points, e.g., frication midpoint or vowel onset. However, the spectra of sibilants vary temporally; thus, it is worth considering how their spectral dynamics vary across vocalic contexts. Vowel context effects were investigated with respect to the trajectories of three psychoacoustic measures computed across the timecourse of native, adult speakers' productions of word-initial, pre-vocalic sibilants. Psychoacoustic spectra were computed from 10 ms windows, spaced evenly across the frication, by passing each window through a bank of gammatone filters, which modeled the auditory system's differential frequency selectivity. From these psychoacoustic spectra, the peak frequency, excitation-drop (difference between maximum high-band and minimum low-band excitation), and the half-power bandwidth of the peak were computed. In Japanese, the peak frequency and the excitation-drop trajectories showed effects of vowel height—the trajectories for high (vs. mid and low) vowel contexts diverging from 50–75% of the fricative duration onward. In English, the excitation-drop trajectories showed similar effects of vowel height; however, the trajectories for high vowels diverged later than in Japanese. Peak bandwidth exhibited context effects only for Japanese /ɕ/, where it was lower in back vowel contexts across the first 75% of the fricative duration.

4aSC13. Phonetic context effects in second-language phonetic category discrimination. Dave C. Ogden (Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan Ave., Ann Arbor, MI 48109, ogdend@umich.edu)

In an exemplar-based model, effects of linguistic experience in L2 acquisition are explained as the conspiracy of experienced tokens in reshaping phonetic category representations (Goldinger, 1998; Johnson, 1997). Such a model predicts that acquisition of a new category is facilitated in phonetic contexts, which are highly dissimilar to L1 experience. This prediction was tested for English-speaking learners' acquisition of the prevoicing-short lag bilabial stop contrast in French. In a phoneme-monitoring task, French learners heard French CV words and responded whenever they heard "the sound p." Word-initially, [p] exemplifies the category /b/ in English but /p/ in French. French vowels included English-like [i u e o] and front rounded non-English [y ø]. If learners' stop category representations retain experienced stops in context, stop tokens preceding [y ø] fall only in the French pre-voicing [b] to short-lag [p] VOT range, while stops preceding [i u e o] include tokens of both languages' categories ([b], [p], and [p^h]). This should lead to faster identification of unaspirated [p] as an instance of /p/ before French-only [y ø] than before shared vowels. Identification of [p] as /p/ was significantly faster before French-only vowels; representations of phonetic segments appear to include information about phonetic context.

4aSC14. Cross-linguistic spectral and voicing properties of [v]. Christina Bjorndahl (Linguist, Cornell Univ., 203 Morrill Hall, 159 Central Ave., Ithaca, NY 14853-4701, cjm295@cornell.edu)

This study reports on the spectral characteristics and voicing properties of [v] in English, Greek, Russian, and Serbian, specifically testing whether inventory structure and the phonological status of [v] correlate with phonetic differences. Phonologically, [v] patterns with obstruents in Greek and English, with sonorants in Serbian and with both obstruents and sonorants in Russian. The present study expands upon previous work in two ways. First, it compares the results to English, which differs from the other three languages in having [w] against which [v] contrasts. Second, it tests the hypothesis that spectral differences correlate with the degree of devoicing. Our results show that [v] is fully and consistently voiced throughout its duration in Greek, Russian, and Serbian, regardless of environment, but exhibits significant devoicing in all environments in English, and moreover varies greatly across speakers. These results show first that the correlation between the spectral properties and phonological status of [v] in Greek, Russian, and Serbian does not arise due to differences in the degree of devoicing of [v]. Second, they suggest that when phonological status is the same, as in Greek and English, other factors such as inventory structure may interact with the phonetic realization of a segment.

4aSC15. The production of Korean coronal obstruents by inexperienced English-speaking learners of Korea. Hanyong Park (Linguist, Univ. of Wisconsin-Milwaukee, Curtin Hall 523, P.O. Box 413, 3243 N. Downer Ave., Milwaukee, WI 53211, park27@uwm.edu)

English-speaking learners of Korean often experience difficulty in learning Korean laryngeal categories (i.e., fortis, lenis, and aspirated). We examine whether the degree of difficulty in producing these laryngeal categories generalizes across the whole coronal obstruent system (i.e., stops and fricatives) and two different prosodic locations (i.e., CV and VCV). In our experiment, English-speaking college students taking first semester Korean course read a list of Korean frame sentences with target stimuli. The target stimuli consist of three coronal stops /t t' t^h/ and two coronal fricatives /s s'/ combined with the vowel /a/ in CV and VCV. To assess production accuracy, native speakers of Korean are asked to identify the consonants from the learners' productions. Results indicate that lenis categories are the most difficult across stops and fricatives in the two prosodic locations, suggesting that the learners develop production skills for laryngeal contrast applicable across the entire obstruent system. However, the degree of difficulty patterns are not similar in CV and VCV: the learners were more accurate producing /t^h/ in VCV than in CV. Thus, these findings suggest that the learners develop the production skills separately for different prosodic locations.

4aSC16. Articulatory targets for coronals in Taiwan Mandarin: A study of EMA, palatography, and linguagraphy. Wei-rong Chen and Yuen-chin Chang (National Tsing Hua Univ., 2F-5, No.62, Ln. 408, Zhong-Hua Rd., Zhubei City, Hsinchu County 302, Taiwan, waitlong75@gmail.com)

Coronal consonants hold a special status for their crowded space for articulatory contrasts. Mandarin and its variants are known to have rich inventories of coronal consonants, where a three-way coronal place contrast is usually maintained. Nonetheless, previous studies have reported a (partial) neutralization of coronal places of articulation in Taiwan Mandarin, while other acoustic studies contend that the three-way contrast remains. The aim of this study is to evaluate the purported coronal neutralization from an articulatory perspective. Five native speakers of Taiwan Mandarin participated in a set of electromagnetic articulographic (EMA), palatographic and linguagraphic experiments in order to investigate the articulatory targets of coronal consonants in Taiwan Mandarin. Our articulatory results are in general consistent with the acoustic results in previous studies, in that the three-way coronal place contrast does exist in Taiwan Mandarin, and the so-called "retroflexes" in Taiwan Mandarin were produced with

constrictions being at "alveolar" zone, rather than "postalveolar" zone as in Beijing Mandarin.

4aSC17. Articulatory similarity in rhotic sounds: A cross-linguistic comparison. Suzanne Boyce, Sarah M. Hamilton, Ahmed Rivera Campos, and Varsha Nair (Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., 345b, Cincinnati, OH 45267, boycese@ucmail.uc.edu)

The question of what phonetic quality defines rhotics as a natural class has been debated. Clinical reports indicate that in many languages, rhotics are developmentally late to emerge and subject to errors that are resistant to remediation, suggesting that rhotics may be distinguished by "complexity in articulation." In American English, the complexity of articulation may derive from its doubly articulated nature; simultaneous palatal and pharyngeal tongue constrictions are consistent features across variants. For other languages with a rhotic liquid, however, it is unclear if this pharyngeal constriction gesture is also an articulatory feature. Using ultrasound, this study compares rhotics from different language families to describe the presence or absence of pharyngeal constriction. Results indicate that pharyngeal constriction is observed across languages in rhotic allophones that have been described as difficult for children to acquire.

4aSC18. A kinematic analysis of Japanese /r/: Effects of syllable position and vowel context. William F. Katz (Commun. Sci. and Disord., Callier Ctr. for Commun. Disord., Univ. of Texas at Dallas,1966 Inwood Rd., TX 75235, wkatz@utdallas.edu), Sonya Mehta (Commun. Sci. and Disord., Univ. of Texas at Dallas, Plano, TX), and Amy Berglund (Commun. Sci. and Disord., Univ. of Texas at Dallas, Dallas, TX)

Japanese /r/ is claimed to vary substantially as a function of context. Greater stricture is thought to occur after pauses and at initial syllable position when compared with intervocalic position. The environment of certain vowels is thought to result in changes to the degree of lateralization of /r/. However, few studies have investigated the articulatory processes involved in realizing the varieties of Japanese /r/ production. To address this issue, we obtained speech recordings for ten native talkers of standard Japanese using an Electromagnetic Articulography (EMA) system. Each talker produced 12 repetitions of /ra/, /ri/, /ru/, /re/, /ro/ in a carrier phrase designed to contrast syllable boundaries: /sore wa VCV VCV desu # CV/ (e.g., "This is *ara ara* [pause] *ra*"). Kinematic data were recorded with sensors placed on the tongue tip (TT), tongue body (TB), tongue left lateral (TL), tongue right lateral (TR), and lower lip (LL) positions. Results to date suggest syllable-initial (post-pausal) /r/ has the greatest stricture, followed by the first VCV production. Productions also showed vowel-dependent differences, with /a/ and /o/ showing noticeable lateral movement patterns. The kinematic data will be compared with perceptual ratings of /r/ quality by native English speakers.

4aSC19. Acoustic and articulatory characteristics of "weak" and "strong" initial sibilants in Chamalal (Andic). Sven Grawunder (Dept. of Linguist, Max Planck Inst. for Evolutionary Anthropology, Deutscher Platz 6, Leipzig 04103, Germany, grawunder@eva.mpg.de) and Zaynab Alieva (Dagestanian State Pedagogical Univ., Makhachkala, Russian Federation)

In the phonological descriptions of a number of Caucasian languages appears a "weak" vs. "strong" contrast for otherwise voiceless obstruents, which was previously described as a lenis-fortis contrast, but recently attributed as geminate. Hence, we investigate field and lab data from the Gigatli dialect of Chamalal (CJI), a language belonging to the Andic branch of Avar-Andic-Tsezic group in the Nakh-Dagestanian family, spoken in Dagestan (Russian Federation). The weak-strong contrast in Chamalal involves glottalic and pulmonic fricatives at the same place of articulation. Concretely the series of central alveolar sibilant fricatives (/s/, /sː/, and /sː'/) is focused, of which all can occur in initial position. Sagittal ultrasound-articulatography of one female speaker demonstrates wide consistency in place of articulation of the tongue tip, although differences in pre-dorsal curvatures between /sː'/ and /s, sː/ are observed. The acoustic data are based on word list elicitations of six speakers (two males/four females): the ejectives show pronounced central envelope peaks and highest intensity slopes of the following vowel, but take a middle position in duration. Two distant peaks are frequently observed in the longer envelopes of "strong" /sː/. /s/ and /sː'/ differ strongest for (>2kHz) weighted COG, whereas /sː/ ranges widely in between.

4aSC20. Dental, or retroflex, that is the question: A study of Mina stop consonants. Alexandra Abell and Steven M. Lulich (Speech and Hearing Sci., Indiana Univ., 200 South Jordan Ave., Bloomington, IN 47405, alabell@indiana.edu)

Mina (also commonly known as Gen) is a language spoken in Togo and the Mono department of Benin. Mina remains little studied in comparison to the related language Ewe. The present study attempts to shed light on Mina phonetics. In particular, the question whether Mina voiced coronal stops form a single category or two categories distinguished by retroflexion is investigated through acoustic, articulatory, and perceptual analyses.

4aSC21. Acoustic evidence for incomplete neutralization of coda obstruents in Korean. Kyounghee Lee, Ann Bradlow, and Matt Goldrick (Linguist Dept., Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, kyoungheelee2013@u.northwestern.edu)

The phonetic manifestation of phonological contrast neutralization is of long-standing interest for our understanding of the phonetic realization of phonological structure. Korean phonology is particularly interesting in this regard with three dimensions of coda contrast neutralization: laryngeal, manner, and palatal. Extending previous work (Kim & Jongman, 1996) that documented complete neutralization of manner contrasts, the present study investigated all of the three types of coda neutralization in both spontaneous (orthography absent) and read speech (orthography present). The results revealed no acoustic differences between most pairs of coda obstruents subject to manner or palatal neutralization. However, laryngeal neutralization was phonetically incomplete with lenis codas (/p, t, k/) differing from their non-lenis counterparts (/p^h, t^h, k'/). In spontaneous speech, this was observed in preceding vowel duration, voicing into closure, and closure duration; in read speech, voicing into closure, closure duration, and release burst duration of a following consonant differed. Our new findings demonstrate that laryngeal coda neutralization in Korean is phonetically incomplete regardless of the presence or absence of orthography. As a case of neutralization beyond final devoicing, this study extends the empirical basis of linguistic theory, discriminating between the predictions of traditional theories versus more recent ones such as exemplar-based theory.

4aSC22. An acoustic study of pulmonic and ejective stops in the final coda position in Yucatec. Masaki Noguchi (Linguist, Univ. of Br. Columbia, 4449 West 14th Ave., Vancouver, British Columbia V6R2Y2, Canada, mskngch@alumni.ubc.ca)

In this study, I present an acoustic analysis of contrast between pulmonic and ejective stops in the phrase final coda position in Yucatec, a Mayan language. Ejective consonants are found in about 16% of the world's language, and there are a relatively large number of acoustic studies about these sounds. However, a significant empirical shortcoming of the previous studies is their exclusive focus on the acoustic realization of ejective consonants in the syllable onset position. This is partly due to a cross-linguistic tendency that ejective consonants are prohibited to occur in the syllable coda position. Therefore, data from Yucatec, which allows ejective consents to occur in the syllable coda position, fill the empirical gap. This study compared pulmonic stop /k/ and ejective stop /k'/ in the phrase final coda position taking a number of acoustic measurements. The results showed (1) stop release is shorter for the ejective, (2) stop closure duration is shorter for the ejective, (3) stop release noise intensity is higher for the ejective, and (4) the contrast between pulmonic and ejective stops affects the voice quality of the preceding short vowel (H1-H2 is lower for the short vowel preceding the ejective).

4aSC23. Uptalk in Southern Californian English and Mexican Spanish. Karla Y. Barranco Márquez (Linguist, UC San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0108, kbarranc@ucsd.edu)

"Uptalk" is the tendency for declarative sentences to end with rising intonation instead of falling intonation. In Southern Californian English, uptalk is realized with a shallow pitch rise [L-H% in Tones and Break Indices (ToBI) labeling]. In contrast to this are polar questions, which are typically realized with steep rises (H-H%) [Ritchart and Arvaniti 2014]. Uptalk has also been impressionistically found in the variety of Spanish spoken in Mexico City. Mexican Spanish has three forms of phrase-final rises: low-to-high rises (LH%), low-to-mid rises (LM%), and high-to-high rises (HH%) [de-la-Mota *et al.* 2010], whereas English only has low and high rises [Beckman & Ayers Elam 1997]. Given the larger inventory of possible phrase-final melodies in Spanish vs. English, uptalk may be realized differently in the former than the latter. To determine whether uptalk in Spanish differs from uptalk in English, we analyzed the intonation from one representative female speaker of each language. Fifty phrases from each language were extracted from recordings of two television shows, *Laguna Beach* (for English) and *Rebelde* (for Spanish). Two coders then labeled each phrase using ToBI conventions for English and Spanish. The phonological and phonetic differences between English vs. Spanish uptalk will be discussed.

4aSC24. Sentence recognition in temporal modulated noise for native and non-native listeners: Effect of language experience. Jingjing Guan, Chang Liu (Univ. of Texas at Austin, 2504A Whitis Ave., Austin, TX 78712, jane.guan@utexas.edu), Sha Tao, Mingshuang Li, Wenjing Wang, and Qi Dong (State Key Lab. of Cognit. Neurosci. and Learning, Beijing Normal Univ., Beijing, China)

Speech recognition in adverse listening conditions is more difficult for non-native listeners than native listeners. Previous work in our laboratories found that Chinese-native listeners with native English exposure may improve the use of temporal cues of noise for English vowel identification. The purpose of this study was to investigate whether such benefits of using temporal modulation in noise were also presented in sentence recognition. IEEE sentence recognition in quiet, stationary, and temporally modulated noise were measured for American English native (EN) listeners, Chinese-native listeners in the United States (CNU), and Chinese-native listeners in China (CNC). Results showed that in general, EN listeners outperformed the two groups of CN listeners in quiet and noise, while CNU listeners had better scores of sentence recognition than CNC listeners. Moreover, at low SNRs, the masking release on sentence recognition from the temporal modulation in noise was greatest for EN listeners, and the smallest for CNC listeners with the CNU listeners in between, while at middle and high SNRs, there was no significant group effect on the masking release, consistent with the findings of English vowel perception. The group difference in using temporal modulation of noise may be associated with acoustic differences between Chinese and English speech.

4aSC25. Listener sensitivity to English- and Spanish-specific sounds in a language categorization task. Cynthia P. Blanco (Linguist, Univ. of Texas at Austin, 305 E. 23rd St., B5100, Austin, TX 78712, cindyblanco@utexas.edu), Colin Bannard (Psych., Univ. of Liverpool, Liverpool, United Kingdom), and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Austin, TX)

Listeners are able to distinguish native from non-native speech with as little as 30 ms of input (Flege, 1984; Kondaurova & Francis, 2008), suggesting that listeners' mental representations include fine-grained acoustic detail about typical and atypical pronunciations in their native language. The current study considers the relative contribution of different language-specific sounds to listeners' language categorization. English monolinguals and early and late Spanish-English bilinguals categorized nonce words containing a language-specific target segment as belonging to English or Spanish. Target segments included the English phonemes /θ, ɹ/, the Spanish phoneme /r/, and the Spanish and English versions of /l, u/, which exist in both languages but vary in their phonetic implementation. All listeners accurately categorized stimuli with /ɹ, r/ and the Spanish pronunciations of /l, u/. English /θ/ and /l, u/ variants received more mixed responses. For these more difficult sounds, late bilinguals were more accurate than the other groups. Early bilinguals responded faster than the other groups to stimuli with language-specific phonemes, and both bilingual groups responded more quickly than monolinguals to /l, u/. The results reveal that the bilinguals' experience with both languages enhances their sensitivity to language-specific cues and may lead to more detailed sound representations for their languages.

4aSC26. Word learning by naïve non-native speakers may not be predictable by cognitive test scores. Irina A. Shport (English, Louisiana State Univ., 260-G Allen Hall, Baton Rouge, LA 70803, ishport@lsu.edu)

The role of cognitive abilities in learning of words with two types of novel sounds—lexical tones and coronal consonants—was compared. Individual differences were expected to play a larger role in implicit learning of tones than consonants. Twenty-two English speakers took tests assessing working memory, attention, executive function, and processing speed (NIH & Northwestern University, 2006–2012, *NIH Toolbox: Cognition*). Then, they learned twelve Vietnamese words varying in their initial consonants [m, t, ʈ, ʐ] and tones (high level, falling, and rising). After training with feedback, the learning was assessed in a 12-word alternative identification test ($N = 792$). The range of accuracy scores was 11.1–80.6%, with the mean accuracy of 43.6% ($SD = 18.5$). The proportion of tonal errors was significantly larger than the proportion of consonantal errors [$\chi(1) = 174.68$, $p < 0.001$]. Mixed-effect regression analyses showed, however, that cognitive test scores did not explain the variance in the word identification accuracy. These results suggests that although lexical tones are more challenging than consonants for speakers of non-tone languages, adult beginning learners may be able to learn words with different types of novel sounds equally efficiently, regardless of individual differences in cognitive abilities.

4aSC27. The comparison of auditory capacity between bilinguals and monolinguals by consonant–vowel dichotic test. Lina Motlagh Zadeh (Commun. Sci. and Disord., Univ. of Cincinnati, 3239 Bishop St., Apt #4, Cincinnati, OH 45220, motlagla@mail.uc.edu), Leila Jalilvand Karimi (Audiol., Shahid Beheshti Medical Sci. Univ., Tehran, Iran), and Noah Silbert (Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

The purpose of the present study is investigation of differences in auditory capacity between bilinguals and monolinguals using the consonant–vowel (CV) dichotic perception test. Bilingualism is one well-studied in psychology and linguistics, with a number of studies revealing that exposure to two languages can lead to the changes in the central system. Auditory capacity refers to the ability to relay information in sound patterns to higher brain centers, with observed measures of auditory capacity reflecting the maximum amount of information that can be processed by the auditory system. The present study probed auditory capacity in 80 normal individuals, 40 bilinguals and 40 monolinguals. Members of the bilingual groups spoke either Turkish or Kurdish from birth and began learning Persian as a second language at or before 6 years of age. Listeners identified distinct consonants presented to each ear in a CV dichotic perception test. Consonant identification accuracy served as a measure of auditory capacity of individual listeners. Results indicated that auditory capacity was greater in the bilingual group. In general, higher scores were gained by bilinguals relative to monolinguals. Due to the large number of bilinguals, knowledge of the similarities and differences between bilinguals and monolinguals and investigation of the effects of second language on central auditory processing are important.

4aSC28. Accounting for multicompetence and restructuring in the study of speech. Charles B. Chang (Dept. of Romance Studies, Boston Univ., 621 Commonwealth Ave., Boston, MA 02215, cbchang@post.harvard.edu)

Phonetic studies meant to generalize to monolingual speakers of a target language have often examined individuals with considerable experience using another language, such as the immigrant native speaker. This paper presents, first, results from a meta-analysis of the literature, suggesting that conflation of ostensibly bilingual ("multicompetent") individuals with monolinguals remains common practice and, second, longitudinal data on speech production that demonstrate why this practice is problematic. Adult native English speakers recently arrived in Korea showed significant changes in acoustic properties of their English production during their first weeks of learning Korean ("phonetic drift") and, furthermore, continued to show altered English production a year later, months after their last Korean class and without extensive use of Korean in daily life. These patterns suggest that the linguistic experience associated with residence in a foreign language environment tends to induce and then prolong phonetic drift of the native language, making the multicompetent native speaker living in a foreign language environment unrepresentative of a monolingual in the native language environment. The speed and persistence of these effects highlight the need for language researchers to be explicit about the population under study and to accordingly control (and describe) language background in a study sample.

4aSC29. Audience design in non-native speech. Jenna T. Conklin, Ashley Kentner, Wai Ling Law, Mengxi Lin, Yuanyuan Wang, and Olga Dmitrieva (Linguist Program, Purdue Univ., 100 North University St., West Lafayette, IN, wlaw@purdue.edu)

Speech has been shown to accommodate the communicative needs of listeners, for example, for increased intelligibility compared to normal speech. Previous research shows that native speakers adapt their speech in the presence of noise (Garnier *et al*., 2006), and when addressing children (Biersack *et al*., 2005) or foreigners (Scarborough *et al*., 2007). However, our knowledge of how *non-native* speakers modify their speech depending on the interlocutor is limited. The goal of this study is to identify acoustic features of non-native speech, which may be affected by a change in listener characteristics, particularly, in terms of language background. Native speakers of Mandarin from the same dialectal area gave directions in a map task to three confederate participants: another native speaker of Mandarin (a non-native speaker from the same L1 background), a native speaker of English (a native speaker of participants' L2), and a native speaker of Russian (a non-native speaker from a different L1 background). Acoustic variables associated with audience design in speech, including measures of speech rate, pitch, vowel duration, vowel quality, and vowel space were examined and compared to results of a detailed survey of participants' language experience and attitudes toward their first and second languages.

4aSC30. Phonetic drift in a first language dominant environment. Wendy Herd, Robin Walden, Whitney Knight, and Savana Alexander (MS State Univ., 2313 Lee Hall, PO Box E, MS State, MS 39762, wherd@english.msstate.edu)

Phonetic drift, changes in the first language (L1) sound system as a result of acquiring a second language (L2), has been documented in learners immersed in L2-dominant environments. Less attention has been given to phonetic drift in speakers learning an L2 in L1-dominant environments. In this study, we conducted a cross-sectional analysis of English learners of Spanish in the United States at beginning ($N = 12$), intermediate ($N = 12$), advanced ($N = 9$), and near native ($N = 6$) proficiency levels. Participants were recorded reading a pseudo-randomized list of words including *heed*, *hayed*, *who'd*, *hoed* to measure drift in vowels and *poll*, *bowl*, *toll*, *dole*, *coal*, *goal* to measure drift in oral stops. Significant differences in vowel quality and in the VOT of voiced stops were found. Intermediate, advanced, and near native learners of Spanish produced vowels in more peripheral positions of the vowel space than beginning learners of Spanish. Similarly, intermediate, advanced, and near native learners produced voiced stops with more negative VOTs than beginning learners. All of these effects were strongest in near native learners. These results suggest that phonetic drift occurs not only when learners are immersed in L2-dominant environments but also as a result of language instruction in L1-dominant environments.

4aSC31. Adaptation to accent affects categorization, but not basic perceptual representation. Matt Lehet and Lori L. Holt (Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Baker Hall, Pittsburgh, PA 15213, mil@andrew.cmu.edu)

Listeners rapidly reweight the mapping of acoustic cues to speech categories in response to local changes in short-term input [Idemaru & Holt, 2011]. For instance, when encountering an accent that reverses the typical correlation of acoustic cues to speech category membership. Here, we examined the level at which this rapid learning occurs. Listeners incidentally encountered an artificial accent in Blocks 1 and 3 of a vowel categorization task. In these blocks, the typical relationship between spectral quality and duration was reversed for /ɛ/ and /æ/ (*setch, satch*) such that /ɛ/ had longer durations than /æ/. Participants down-weighted their reliance on the duration of spectrally ambiguous vowels after brief exposure to the accent. They rapidly returned to using duration when the English correlation was restored (Block 2). Additionally, we tested whether vowel duration exerts a durational contrast effect on adjacent consonants and, if so, whether this effect was modulated by perceptual down-weighting of duration in Blocks 1 and 3. We observed consistent durational contrast effects across blocks. Even when participants down-weighted vowel duration for categorization, vowel duration exerted a durational contrast effect on the following consonant. Learning leaves the basic perceptual representation intact. We describe a model of the results.

4aSC32. Contributions of practice with feedback and testing without feedback to learning of a non-native phonetic contrast. Beverly A. Wright, Emma K. LeBlanc, Jessica S. Conderman, and Courtney S. Coburn (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60202, b-wright@northwestern.edu)

Listeners learn non-native phonetic contrasts given practice and feedback, but what is the dose-response curve for that training? Does testing without feedback affect that response? We trained native English speakers on a non-native phonetic contrast between pre-voicing and voicing on two consecutive days, varying the number of daily training trials across groups. Practice with feedback for 240, 120, or 60 trials/day all led to a similar steepening of the category-boundary slope, but practice with feedback for 30 trials/day yielded no learning. To determine whether the daily pre- and post-training tests without feedback (120 trials/test) affected this outcome, we trained another group 60 trials/day with feedback, but removed all but the final post-test. This group did not learn, indicating the tests without feedback facilitated improvement. Finally, to establish whether the benefit from these tests came from task performance, we trained another group 60 trials/day with feedback, but replaced the original tests with stimulus exposure alone until the final post-test. This group's ending performance resembled that obtained with task performance during the tests without feedback. Thus, trials with feedback can function as an all-or-none trigger for recruiting the contribution of trials without feedback, or mere stimulus exposures, to speech learning. [NIH supported.]

4aSC33. Neurophysiological correlates of perceptual learning of Mandarin Chinese lexical tone categories: An event-related potential study. Guannan Shen and Karen Froud (Teachers College, Columbia Univ., 509 West 122nd St., Apt. 18, New York, NY 10027, mandy.g.shen@gmail.com)

Recent studies have shown that certain brain event-related potentials (ERPs) are sensitive to auditory perceptual categorical boundaries. This study investigates brain responses to lexical tone categorization for three groups of adult listeners: (1) native English speakers who had no exposure to Mandarin before age 17, but took advanced Mandarin courses as adults; (2) naïve English speakers; and (3) native Mandarin speakers. Two tonal continua were derived from natural speech through interpolation within two tonal contrasts (Tone 1/Tone 4; Tone 2/Tone 3). First, category boundaries were examined through classic identification and discrimination tasks. Secondly, high-density electroencephalography (EEG) was used to record brain responses while participants listened to tones in two oddball paradigms: across-category and within-category. If perception of lexical tones is categorical, cross-category deviants are expected to elicit larger ERP responses (specifically, mismatch negativity (MMN) and P300) than within-category deviants. Both behavioral and ERP results indicate that lexical tones are perceived categorically by native Chinese speakers but not by inexperienced English speakers. Although English learners of Chinese demonstrated categorical perception in behavioral tasks, their ERP response amplitudes were attenuated, and did not differ between within- and across-category conditions. Acoustic cues and characteristics of L2 phonological learning in adulthood are discussed.

4aSC34. Asymmetries in the perception of Mandarin tones: Evidence from mismatch negativity. Stephen Politzer-Ahles and Kevin Schluter (NYUAD Inst., New York Univ. Abu Dhabi, PO Box 903, New York, NY 10276-0903, spa268@nyu.edu)

The mismatch negativity (MMN) component of the electroencephalogram reflects phonological asymmetries: specifically, greater MMN is elicited by infrequent (deviant) tokens when the features of the frequent (standard) token are phonologically underspecified than when those features are phonologically fully specified. Such asymmetries have not been studied in non-Indo-European languages or in suprasegmental contrasts, which might not be represented in the same way as segments. Therefore, the present study investigated the neural representation of Mandarin contour tones. Mandarin third tone (T3) is realized as second tone (T2) in certain contexts, whereas it has no alternation relationship with fourth tone (T4). The present study examined MMNs elicited by T3~T2, T3~T4, and T2~T4 contrasts (in both directions). Asymmetrical MMN effects were elicited in both T3~T2 and T3~T4 contrasts: MMN was smallest when the standard was T3. The results suggest that T3 has a less specified neural representation than both T2, which it alternates with, and T4, which it does not. This may be due to the featural representation of T3 (in standard Mandarin it is a Low tone, which is typologically less marked) or due to its behavior (i.e., it may be the case that features which alternate always trigger reduced MMN effects).

4aSC35. Asymmetries in vowel perception: Effects of formant convergence and category "goodness". Matthew Masapollo, Linda Polka (McGill Univ., 2001 McGill College, 8th Fl., Montreal, Quebec H3A 1G1, Canada, matthew.masapollo@mail.mcgill.ca), and Lucie Ménard (Univ. of PQ at Montreal, Montreal, Quebec, Canada)

The mechanisms underlying directional asymmetries in vowel perception have been the subject of considerable debate. One account—the Natural Referent Vowel (NRV) framework—suggests that asymmetries reflect a *language-universal* perceptual bias, such that listeners are predisposed to attend to vowels with greater formant convergence (Polka & Bohn, 2011). A second (but not mutually exclusive) account—the Native Language Magnet (NLM) theory—suggests that asymmetries reflect an experience-dependent *language-specific* bias favoring "good" exemplars of native language vowel categories (Kuhl, 1993). We tested the above hypotheses by investigating whether listeners, from different language backgrounds, display asymmetries influenced by formant proximity and/or language experience. Specifically, we examined monolingual English and French listeners' performance in a within-category AX vowel discrimination task, using variants of /u/ that systematically differed in *both* their degree of formant proximity (between F1 and F2) and category "goodness" judgments. Results revealed asymmetries that pattern as predicted by NRV when pairs of /u/ tokens exhibited a relatively larger difference in their F1-F2 convergence patterns, and as predicted by NLM when the pairs of /u/ tokens exhibited a relatively smaller difference in their F1-F2 convergence patterns. These findings suggest that language-universal perceptual biases and specific language experience interact to shape vowel perception.

4aSC36. Nonnative phonetic category training in varying acoustic environments. Eleni L. Vlahou, Aaron Seitz (Psych., Univ. of California, Riverside, 900 University Ave., Riverside, CA 92521, evlahou@gmail.com), and Norbert Kopčo (Inst. of Comput. Sci., P. J. Šafárik Univ., Košice, Slovakia)

Past research has shown that reverberation has a pronounced detrimental effect on speech intelligibility, but no previous studies investigated how it affects the acquisition of novel phonetic categories. Here, we present a follow-up to a previous study that examined adult nonnative phonetic learning using speech stimuli presented in reverberant environments [Vlahou *et al.* (2014), ARO Abstract #806]. Listeners were trained to discriminate a novel dental-retroflex contrast in Hindi. Using virtual acoustics, stimuli were presented in anechoic space, in a single room, or in multiple rooms. For some subjects, training was supervised, consisting of a 2AFC categorization task with trial-by-trial feedback. For other subjects, training was unsupervised, in the form of a videogame which promoted stimulus-reward contingencies. Performance was evaluated on trained and untrained sounds presented in familiar and unfamiliar rooms. Overall, trained listeners outperformed untrained listeners. Supervised training induced more robust learning of the trained material and generalization of learning to untrained voice. Introducing multiple reverberant environments was beneficial for unsupervised, but not for supervised, training. These results suggest that phonetic adaptation to reverberation is robust and that experiencing novel phonemes in various acoustic environments can enhance unsupervised phonetic category learning. [Work supported by APVV-0452-12, TECHNICOM ITMS: 26220220182, and NSF-BCS-1057625.]

4aSC37. English listeners use suprasegmental lexical stress online during spoken word recognition. Alexandra Jesse (Psychol. and Brain Sci., Univ. of Massachusetts, Amherst, MA), Katja Poellmann, and Ying-Yee Kong (Commun. Sci. & Disord., Northeastern Univ., 360 Huntington Ave., 226 FR, Boston, MA 02115, yykong@neu.edu)

Lexical stress facilitates spoken word recognition. In English, listeners primarily rely on vowel reduction in unstressed syllables as a cue to lexical stress, but are nevertheless sensitive to suprasegmental cues (Cooper *et al.*, 2002 *Lang. Speech*). In the present study, we tested with a visual-world eye-tracking paradigm whether English listeners utilize suprasegmental stress cues online during word recognition. On each trial, listeners heard one of four displayed printed words. Two of the words (critical words) were segmentally identical in their first two syllables but differed in the suprasegmental realization of stress in their first syllable. The first syllable had either primary (e.g., admiral) or secondary (admiration) lexical stress. The other two words were phonologically and semantically unrelated to the critical items. In the critical word trials, listeners looked more frequently at the targets with initial primary stress than at the competitors with secondary stress during the presentation of the first two syllables. No difference was found when the targets had secondary stress. The degree to which competitors were fixated was not modulated by stress. This suggests that English listeners use the presence, but not the absence, of suprasegmental cues to primary lexical stress during word recognition.

4aSC38. Rule-based and information-integration categorization during an incidental learning task. Casey L. Roark and Lori L. Holt (Psych. Dept., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, croark@andrew.cmu.edu)

Research in visual category learning supports the existence of dual category-learning systems with distinct neurobiological substrates. Rule-based category learning is engaged by category input distributions that vary orthogonally in perceptual space whereas information integration learning is engaged when categories are defined across multiple input dimensions. Investigators have recently begun to explore these dual systems in audition, including phonetic category learning. However, investigations of dual category learning systems have tended to use category training tasks in which participants actively search for category-relevant dimensions, make overt category labeling decisions, and receive explicit feedback about the correctness of their responses. Recent neuroimaging research highlights the fact that overt category training tasks like this engage learning systems that differ from those engaged by more incidental category learning. Since most natural category learning occurs under more incidental conditions, it is important to understand how training tasks interact with category learning systems. The aim of the present research is to investigate the interaction of input distribution and task. We examine nonspeech auditory category learning across the same perceptual input space for rule-based and information-integration categories in incidental and overt training tasks to characterize how training task interacts with input category distributions.

4aSC39. Listeners' sensitivity to nasal coarticulation and its interactions with lexical neighborhood density. Kuniko Nielsen (Linguist, Oakland Univ., 320 O'Dowd Hall, Rochester, MI 48309-4401, nielsen@oakland.edu) and Rebecca Scarborough (Linguist, Univ. of Colorado Boulder, Boulder, CO)

Previous research has shown that words from dense phonological neighborhoods, thus subject to greater lexical competition, are hyperarticulated (Wright, 2004; Munson and Solomon, 2004) and produced with a greater degree of coarticulation (Scarborough, 2013). The current study investigates

listeners' sensitivity to lexically conditioned degree of nasal coarticulation. If the basis of attested Neighborhood Density (ND) effects is listener-oriented (as suggested by Wright), listeners' sensitivity to different degrees of coarticulation may be influenced by ND patterns as well. Forty-three native speakers of American English participated in a forced-choice discrimination task. The stimuli consisted of 32 monosyllabic words with nasal codas, and the degree of vowel nasality was natural or artificially increased or decreased. A Generalized Linear Mixed Effects regression was performed on Correct Response with Direction of Manipulation and ND as fixed factors. The model revealed a significant interaction of Direction and ND ($p < 0.001$), where increased nasality was discriminated more correctly among high ND words while decreased nasality was discriminated more correctly among low ND words. This result shows that listeners are indeed sensitive to differences in degree of nasal coarticulation, and in ways that reflect neighborhood conditioned patterns.

4aSC40. Perceptual scaffolding of non-native speech categories through videogame-based training. Ran Liu and Lori L. Holt (Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, ranliu@cmu.edu)

Adults learn many new tasks with ease, but acquiring the sounds of a new language is notoriously difficult. Decades of attempts to develop effective training regimens have focused primarily on highly *explicit* approaches to training adults to categorize non-native speech sounds. Participants are aware of the phonetic distinctions they are learning, focus their attention on the contrasts or categories of interest, and are provided with some form of overt trial-by-trial feedback. A growing body of perceptual learning research suggests that directing attention away from the task of learning sounds by embedding training in tasks with overt goals that are unrelated to sound training can lead to efficient learning gains. We leverage this idea to incidentally train native English-speaking adults on non-linguistic auditory categories that are highly relevant to Mandarin Chinese lexical tone perception. We observed significant incidental learning of the nonspeech auditory categories as a result of five days of incidental videogame training and highly significant generalization to natural spoken productions of Mandarin tones. We also observed preliminary evidence that nonspeech category learning transferred to boost Mandarin vocabulary learning. We discuss these results in the context of incidental category learning and perceptual scaffolding of non-native speech categories.

4aSC41. Incidental auditory category learning. Lori L. Holt, Yafit Gabay (Psych., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15232, loriholt@cmu.edu), Frederic Dick (Birkbeck, Univ. of London, London, United Kingdom), and Jason Zevin (Univ. of Southern California, Los Angeles, CA)

Little is known about how auditory categories are learned incidentally—that is, in the absence of overt category decisions or experimenter-provided feedback. This is an important gap, as learning in the natural environment does not arise from explicit feedback and there is evidence that the learning systems engaged by traditional tasks are distinct from those recruited by incidental category learning. We examined incidental auditory category learning with a novel paradigm, the Systematic Multimodal Associations Reaction Time (SMART) task, where participants rapidly detect and report the appearance of a visual target in one of four possible screen locations. Although the overt task is rapid visual detection, a brief sequence of sounds precedes each visual target. These sounds are drawn from one of four distinct sound categories that predict the location of the upcoming visual target. These many-to-one auditory-to-visuomotor correspondences support incidental auditory category learning. Participants incidentally learn categories of complex acoustic exemplars and generalize this learning to novel exemplars and tasks. Further, learning is facilitated when category exemplar variability is more tightly coupled to the visuomotor associations than when the same stimulus variability is experienced across trials. We relate these findings to theories of incidental phonetic category learning.

Session 4aSP

Signal Processing in Acoustics: Beamforming, Optimization, Source Localization, and Separation

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Contributed Papers

9:00

4aSP1. Information-theoretic optimization of multiple-sensor positioning for passive narrowband acoustic source localization. Thomas J. Hayward and Mitchell A. Potter (Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375, thomas.hayward@nrl.navy.mil)

Fundamental information-theoretic quantities, including conditional entropy of source location given received complex spectral values and per-iteration information gain (relative entropy) are applied as performance metrics to the optimization and real-time adaptation of receiver array spatial configurations for iterative (sequential) Bayesian localization of narrowband acoustic sources. Computational examples illustrate the application of these performance metrics to the adaptation of mobile-sensor spatial positioning and to the optimization of element positions for fixed vertical arrays in a shallow-water waveguide. Evolutionary search algorithms [Bäck and Schwefel, Evolutionary Computation **1**(1), 1–23 (1993)] are investigated as a unified computational approach to both optimization problems. The optimized array spatial configurations are compared with configurations optimized with respect to traditional (energy-based) performance metrics, and the differences are interpreted. [Work supported by ONR.]

9:15

4aSP2. On the estimation of bearing using a single hydrophone. Edmund Sullivan (Res., prometheus, 46 lawton brook Ln., Portsmouth, RI 02871, bewegungslos@fastmail.fm)

It is well known that a single moving hydrophone can estimate the bearing of a pure tone source if the source frequency is known *a priori*. It is also well known that with two or more hydrophones, a moving array can provide an improved estimate of bearing by jointly estimation the source frequency and bearing with a Kalman filter. This is due to the introduction of the phase information contained in the Doppler. This leads to the question of whether by jointly estimating the bearing and source frequency with a Kalman filter, a single hydrophone will suffice. By using a form of the observability matrix adapted to the nonlinear case, it is shown that in general, it is not possible. The reason is shown to be related to the fact that the bearing information in the Doppler is not sufficient and the introduction of a spatial phase, by introducing at least one more hydrophone is necessary.

9:30

4aSP3. The effect of ground sensor and UAV flight path geometry on the tomographic reconstruction of a weakly sheared daytime convective planetary boundary layer. Anthony Finn and Kevin Rogers (Defence & Systems Inst., Univ. of South Australia, Bldg. W, Mawson Lakes, SA 5095, Australia, anthony.finn@unisa.edu.au)

A signal analysis of the sound of a propeller-driven unmanned aerial vehicle (UAV) shows that its acoustic signature comprises a set of strong narrowband tones superimposed onto a broadband random component. If such a UAV overflies an array of microphones, the projected and observed Doppler shifts in frequency of the narrowband tones may be compared and converted into effective sound speed values: 2- and 3D spatially varying atmospheric temperature and wind velocity fields may then be estimated using tomography. The technique has practical application in a number of research fields. In this paper, we examine the influence of UAV flight path and ground sensor geometry on the feasibility and usefulness of UAV-based atmospheric tomography. Realistic conditions for a weakly sheared daytime convective atmospheric boundary layer are synthesized through use of massively parallel large eddy simulation code that utilizes pseudo-spectral differencing in horizontal planes and solves an elliptic pressure equation. Particular attention is paid to the accuracy with which the surface layer (lowest 50 m of atmosphere) may be reconstructed using UAV-based acoustic tomography as this region typically experiences the greatest spatio-temporal variation in temperature and wind speed; and arrangements of UAV flight path and sensor geometry do not permit ray paths to intersect without the UAV flying very low and disturbing the atmosphere. The influence of meteorological observations obtained onboard the UAV and by ground sensors is also examined.

9:45

4aSP4. Azimuth-elevation direction finding using one four-component acoustic vector-sensor spread spatially along a straight line. Yang SONG (Dept. of Elec. Eng. and Information Technol., Universität Paderborn, Paderborn, North Rhine - Westphalia, Germany) and Kainam T. Wong (Dept. of Electron. & Information Eng., Hong Kong Polytechnic Univ., DE 605, Hung Hom KLN, Hong Kong, ktwong@ieee.org)

A considerable literature has developed to use the four-component acoustic vector-sensor to estimate the azimuth-elevation direction-of-arrival of incident acoustic sources. A four-component acoustic vector sensor (a.k.a. a "vector hydrophone" in underwater applications) consists of a pressure sensor, plus three identical but perpendicular univariate velocity-sensors, which are often idealized as collocated. One shortcoming of the above acoustic vector sensor is the point-size of its array aperture, due to the spatial collocation of all its four component-sensors. If these four component-sensors are placed at different spatial locations, there would arise spatial phase factors among these component-sensors' data, and the array-manifold of (1) would be invalid, thus the incident wavefield's propagation direction is no longer so straight-forwardly obtainable. However, Song & Wong (JASA **4**, 2013) shows that how the four component-sensors may be allowed to spread out arbitrarily in three-dimensional space (thus sampling the incident wavefield at different locations), while not just achieving direction finding, but to do so with increased accuracy. This paper will focus on one particular array grid to spread out the four component-sensors—on a straight line in any permutation.

10:00–10:15 Break

10:15

4aSP5. Role of adaptive beamforming with random and irregular arrays. Paul Hursky (HLS Res. Inc., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, paul.hursky@hlsresearch.com)

When we design an array to accommodate as many as 4–5 octaves of bandwidth, the design must provide a variety of apertures and element spacings. Further, the concept of operations may be such that no structure is available for maintaining a regular array structure (even if we actually wanted one). One example of this is an array of freely drifting sonobuoys. Clearly, with a limited number of elements (far fewer than would be needed to provide half wavelength spacing for the highest frequency over the aperture needed at the lowest frequency), compromises must be made. Solutions include combinations of nested apertures, geometrically increasing spacings, and random spacings to varying degrees—in short irregular and random arrays. Such designs have very poor conventional beam response patterns, with very high sidelobes, compared to the fully populated arrays spaced at half wavelength. However, these objections are a red herring. We will show that adaptive beamforming produces beam responses with very low sidelobes. We will also show that irregular and random arrays indeed outperform regular arrays (of equivalent cost), if a wide bandwidth and range of interference directions are considered. We will also briefly review requirements for array calibration for adaptive beamforming.

10:30

4aSP6. Compressive beamforming with LASSO. Peter Gerstoft (SIO Marine Phys. Lab. MC0238, Univ. of California San Diego, 9500 Gillman Dr., La Jolla, CA 92093-0238, gerstoft@ucsd.edu) and Angelliki Xenaki (Appl. Mathematics and Comput. Sci., Tech. Univ. of Denmark, Kgs. Lyngby, Denmark)

For a given sound field observed on an array, compressive beamforming reconstructs the direction-of-arrival (DOA) map using a sparsity constraint on the source distribution. The problem is posed as an underdetermined problem expressing the pressure at each receiver as a superposition of plane waves associated with each DOA, a phase delayed sum of source amplitudes. The L1 sparsity constraint makes the problem solvable with convex optimization and the sparsity constraint gives improved resolution. We here derive the sparse source distribution using maximum a posteriori (MAP) estimates for both single and multiple snapshots. Compressive beamforming does not rely on any matrix inversion and thus works well even for single snapshots where it gives higher resolution than conventional beamforming. For multiple snapshots in a free-space environment, compressive beamforming has a performance similar to MVDR. In a multi-path environment MVDR fails, but compressive beamforming works well. The superior resolution of compressive beamforming is demonstrated with vertical array data from event S5 in the SWellEx96 experiment in a multi-path shallow water environment.

10:45

4aSP7. On beyond hidden source estimates—Improving the output of blind source separation algorithms. Richard Goldhor, Joel MacAuslan, and Keith Gilbert (Speech Technol. & Appl. Res., 54 Middlesex Turnpike, Entrance D, Bedford, MA 01730, rgoldhor@sprynet.com)

In the presence of multiple simultaneously active acoustic sources, microphone response signals are typically additive mixtures of "acoustic images"—that is, filtered and delayed versions of hypothetical yet intangible "source signals." Under certain conditions, Blind Source Separation (BSS) algorithms can "demix" sets of such microphone responses into output signals often called "Hidden Source Estimates" and characterized as estimates of those inaccessible source signals. However, despite the name of these algorithms and the characterization of their outputs, BSS algorithms generally do not actually reconstruct the original source signals. Indeed, it is not even clear that the notion of "source signal" is always well-defined. We argue that a valid BSS output can best be understood as an estimate of some unknown member of an entire equivalence class of related signals, and that certain members of these classes are much more useful and ontologically well-grounded than others. Notably, acoustic images are particularly interesting members of these equivalence classes. We present a method for converting arbitrary hidden source estimates into acoustic images, and explore the utility of enhancing BSS algorithms so that they output specific acoustic images rather than generic hidden source estimates.

11:00

4aSP8. Blind source localization in a room based on wavefield separation. Thibault Nowakowski, Julien De Rosny, Laurent Daudet (Institut Langevin - UMR7587 - CNRS, 1 rue Jussieu, Paris 75005, France, thibault.nowakowski@espci.fr), and François Ollivier (UPMC - Institut Jean le Rond D'Alembert, St-Cyr l'Ecole, France)

Narrowband localization of sources in a room is a challenging problem because of the multiple reflections off the walls. Recently, we have been developing methods to localize monochromatic sources within an array of a few tens of microphones, without any knowledge on the physical properties of the room. To that end, we use a wavefield dereverberation approach in which the diffuse part is canceled, thanks to a projection operator built from a plane wave basis. However, this basis requires that no heterogeneities are present in the space of interest between the microphones. To overcome this limitation, in the case of an heterogeneous space, we show that a new projection operator can be experimentally built from a set of measurements of the responses between the microphones and some sources. This projection operator removes the reverberation, and can be used as preprocessing to locate the sources despite the heterogeneities, for instance by using classical beamforming processing. The method is first validated with numerical simulations. Then, experiments are performed in a large room, with an array composed of 100 microphones. A source, emitting at 500 Hz, can be located close to a strong reflector with an accuracy of about 10 cm.

11:15

4aSP9. Blind separation of heart sounds. Lingguang Chen, Sean F. Wu (Dept. of Mech. Eng., Wayne State Univ., 5050 Anthony Wayne Dr., Detroit, MI 48202, fj6467@wayne.edu), Yong Xu (Dept. of Electric and Comput. Sci. Eng., Wayne State Univ., Detroit, MI), William D. Lyman, and Gaurav Kapur (Dept. of Pediatrics, Wayne State Univ., Detroit, MI)

This paper presents blind separation of heart sounds by de-convoluting the directly measured mixed signals using the temporal Green's function or the point source separation (PPS) method (Wu and Zhu, JASA, 2012). The objective of this study is to examine the feasibility of using a practical and cost-effective method to separate the aortic, pulmonic, mitral and tricuspid sounds that are involved in the first and second heart sounds, respectively. It is emphasized that because many parameters such as the locations from which the heart sounds are originated, as well as the speed at which heart sounds travel inside a human body are unknown a priori, it is unrealistic to expect a perfect separation of the heart sounds. To begin with, we conduct a numerical simulation test that uses an iteration scheme to locate sources and determine the speed of sound. This is done by applying PPS algorithm repeatedly under different source locations and speeds of sound. Results show that this iteration always leads to a convergent range of source locations and speeds of sound, from which approximate values of source locations and speed of sound can be determined. Once this is completed, the signals from individual sources are separated by de-convoluting the mixed signals with respect to individual source locations. This blind source separation methodology is then applied to the heart sounds measured on volunteers to separate the aortic, pulmonic, mitral, and tricuspid sounds.

Session 4aUW

Underwater Acoustics: Three-Dimensional Underwater Acoustics Models and Experiments I

Ying-Tsong Lin, Cochair
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Marcia J. Isakson, Cochair
Applied Research Laboratories, The University of Texas at Austin, 10000 Burnet Road, Austin, TX 78713

Chair's Introduction—8:00

Invited Papers

8:05

4aUW1. Three-dimensional propagation in the open ocean: Observations and modeling. Kevin D. Heaney and Richard L. Campbell (OASIS Inc., 11006 Clara Barton Dr., Fairfax Station, VA 22039, oceansound04@yahoo.com)

A primary feature of ocean acoustic propagation is that vertical environmental scales and gradients are generally an order of magnitude smaller than horizontal scales. As such, ocean acoustic experiments and modeling have developed with the view that the in-plane propagation is primary. There are environments, however, where the horizontal variability can be significant enough to lead to a change in the observed acoustic field due to out-of-plane refraction or diffraction. This is particularly evident at very low frequencies, where the open ocean can no longer be considered deep water and the seafloor topography is not negligible. In this paper, several recent observations of the impact of three-dimensional propagation on global propagation will be presented, as well as a fully 3-D parabolic equation, which is efficient enough to handle very long range propagation at low frequencies. The final modeling illustration will be the impact of ocean mesoscale on arrival angle of low frequency long range signals. This is implications to the detection capability of systems like the Comprehensive Testban Treaty Organization.

8:25

4aUW2. Bottom-diffracted surface-reflected arrivals in the Philippine Sea. Ralph A. Stephen (Geol. & Geophys. MS 24, WHOI, 360 Woods Hole Rd., Woods Hole, MA 02543-1592, rstephen@whoi.edu), Peter F. Worcester (Scripps Inst. of Oceanogr., La Jolla, CA), Richard L. Campbell (OASIS, Inc., Lexington, MA), Ilya A. Udovydchenkov (Mitre Corp., Bedford, MA), and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA)

Bottom-diffracted surface-reflected (BDSR) arrivals were first identified in the 2004 Long-range Ocean Acoustic Propagation Experiment (LOAPEX) in the North Pacific (Stephen *et al*., 2013, JASA, **134**, 3307–3317). For sources at long ranges in deep water, they can be observed throughout the water column, but at depths above the conjugate depth they are obscured by ambient noise and energy propagating in the ocean sound channel (PE predicted arrivals). In deep water (deeper than the conjugate depth), ambient noise and PE predicted arrivals are sufficiently quiet that BDSR paths, scattered from small seamounts, can be the largest amplitude arrivals observed. In the Ocean Bottom Seismometer Augmentation in the Philippine Sea (OBSAPS) Experiment in April–May 2011, we observed BDSR arrivals on ocean bottom seismometers at relatively short ranges (less than 50 km). The experiment was designed to further define the characteristics of the BDSRs and to understand the conditions under which BDSRs are excited and propagate. BDSR arrivals are distinct, with little indication of coda or reverberation, and the scattering point appears to be discrete. The BDSR mechanism provides a means for acoustic signals and noise from distant sources to appear with significant strength on the deep seafloor. [Work supported by ONR.]

8:45

4aUW3. The role of water column variability on three-dimensional sound propagation in shallow water. Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu)

Simultaneous measurements of sound field and the sound speed profile perturbed by passage of internal waves in shallow water and the resulting modal evolution in space-time domain is reported. The waveguide modal behavior for broadband signals centered at 100, 200, and 300 Hz with the bandwidth of 60 Hz during the Shallow Water 2006 (SW06) experiment is studied using an L-shaped hydrophone array. Several cases are considered where the angle of the internal wave front with the acoustic track ranges from −8 to + 83 degrees. Mode filtering obtained from the vertical array and tracking the filtered mode along the horizontal array reveals that individual modes propagating between the source and receiver arrive at different angles. Using the method of vertical modes and horizontal rays the phase of the acoustic field is calculated and the change in the angle of arriving ray at the L-shaped array, that is the horizontal angle of refraction, is obtained. A 3D PE model is used for comparison between the calculations and the experimental data. [Work supported by ONR 322OA.]

4a THU. AM

Contributed Papers

9:05

4aUW4. Bathymetric refraction effects associated with marine pile driving. David Dall'Osto (Appl. Phys. Lab. at Univ. of Washington, Seattle, WA) and Peter H. Dahl (Appl. Phys. Lab. and Dept. of Mechanical Eng., Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dahl@apl.washington.edu)

Measurements of underwater noise from vibratory pile driving from a marine construction site in Puget Sound, Washington, are studied using line array-based measurements made at range 16 m from the pile source and single hydrophones at range 417-m on one transect, and at ranges 207-m and 436-m on another transect running approximately parallel to a sloping shoreline. Using adiabatic modes, the field from an incoherent line source model is propagated along these two transects. This approach works well except for the 436-m measurements located on the along-shore transect, where the observed levels for frequencies <300 Hz are significantly lower than predicted. These observations are interpreted in the context of shadow zones associated with bathymetric refraction for a downward sloping beach [Deane and Buckingham, J. Acoust. Soc. Am. **93**, 1319 (1993)]. The modal amplitude due to horizontal refraction is evaluated with the parabolic wave equation, written in terms of the depth dependent phase velocity of each mode [Ballard, Proc. Meetings Acoust.ics **15**, 070001 (2012)]. The results are significant as sloping beaches form many of the boundaries to waters where pile driving occurs, and where protective environment underwater noise monitoring is necessary, such as in Puget Sound.

9:20

4aUW5. Matched mode geoacoustic inversion of broadband signals in shallow water. Hefeng Dong (Dept. Electronics and Telecommunications, NTNU, Norwegian Univ. of Sci. and Technol., Trondheim NO 7491, Norway, hefeng@iet.ntnu.no), Mohsen Badiey (Univ. of Delaware, Newark, DE), and Ross Chapman (Univ. of Victoria, Victoria, British Columbia, Canada)

A matched mode geoacoustic inversion is presented for broadband acoustic signals recorded in experiments at a shallow water region of the New England Bight. First the modal behavior of the waveguide in the presence of 3D effects due to the water column and bottom bathymetry is examined. Spatial measurements of the temperature and salinity profiles and bathymetry provide data for calculating 3D propagation of the broadband acoustic wave field. Frequency shift of the broadband signal along one of the source-receiver paths in the experiment is attributed to internal waves propagating in the water column. Geoacoustic model parameters of the seabed along a source-receiver track with constant bathymetry are estimated by matched mode processing through matching the phase of the mode signals recorded at a Vertical Line Array. A warping transform was applied to identify and isolate modes, and the filtered modes were inverse transformed back to the time domain to reconstruct the mode signals. The inversion results are compared with the results from other inversion methods to assess the performance of the inversion scheme for estimating reliable values of the geoacoustic parameters in the experiment region.

9:35

4aUW6. Horizontal coherence of sound propagation in the presence of internal waves on the New Jersey continental shelf. Lin Wan and Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., 104 Robinson Hall, Newark, DE 19716, wan@udel.edu)

The knowledge of spatial coherence of sound field is required for sonar application and array design. In shallow water waveguides, the spatial and temporal evolution of the temperature field induced by internal waves will cause signal fluctuations and variations in spatial coherence. During the Shallow Water Acoustic Experiment 2006, the three dimensional temperature field of internal waves was measured and reconstructed while acoustic signals were simultaneously transmitted between various sources and an L-array. These internal wave events have been used to investigate the internal wave effect on temporal coherence and vertical structures of acoustic normal modes [Wan and Badiey, J. Acoust. Soc. Am. **135**, 2014]. In this paper, the horizontal structures of acoustic normal modes are examined. The arrival angles of normal modes are analyzed. The horizontal coherence of sound field is obtained at different frequencies for these internal wave events. The relationship between horizontal coherence and internal wave parameters, such as propagation direction, speed, and amplitude, is discussed. [Work supported by ONR322OA.]

9:50–10:10 Break

Invited Papers

10:10

4aUW7. Laboratory scale measurements of three-dimensional sound propagation in the presence of a tilted penetrable bottom. Frédéric Sturm (Ctr. Acoustique, LMFA UMR CNRS 5509, Ctr. Acoustique, Ecole Centrale de Lyon, 36, Ave. Guy de Collongue, Ecully 69134, France, frederic.sturm@ec-lyon.fr), Jean-Pierre Sessarego (LMA, UPR CNRS 7051, Marseille, France), and Alexios Korakas (Ctr. Acoustique, LMFA UMR CNRS 5509, Ecully Cedex, France)

Results of laboratory-scale measurements of long-range across-slope pulse propagation in three-dimensional wedge-shaped oceanic waveguides with a sandy bottom are reviewed. The experimental data considered were collected during two laboratory-scale experimental campaigns that were led in 2006 and 2007 (after a required calibration phase) in the large indoor shallow-water tank of the LMA-CNRS laboratory in Marseille. Operational frequencies and water depths were chosen to produce a reduced number of propagating modes so as to facilitate the analysis of the received signals. The experimental data contain strong 3-D effects like mode shadow zones and multiple mode arrivals, all consistent with well-known three-dimensional effects described in the literature. The data are compared with numerical solutions obtained using a fully three-dimensional parabolic equation based model. Comparisons are performed in both time domain (received signals at distinct ranges and depths) and frequency domain (TL-*versus*-range curves at distinct depths for several discrete frequencies). In both cases, the experimental data are in good agreement with the numerical predictions.

10:30

4aUW8. Measurements of three-dimensional acoustic propagation in a scale-model canyon. 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)

Scale-model acoustic propagation experiments were conducted in a laboratory tank to investigate three-dimensional (3D) propagation effects induced by range-dependent bathymetry. The model bathymetry, patterned after measured bathymetric data, represents a portion of the Hudson Canyon at 1:7500 scale and was fabricated from closed-cell polyurethane foam using a computer-numerically controlled (CNC) milling machine. In the measurement apparatus, a computer-controlled positioning system locates the receiving hydrophone at user-defined points in 3D space while the stationary source hydrophone emits broadband pulsed waveforms. Precise control of the receiving hydrophone permits the creation of synthetic arrays from which horizontal and vertical beamforming is performed. Results are shown for propagation paths along and across the axis of the canyon where the received time series are post-processed to estimate travel times, transmission loss, and horizontal and vertical arrival angles. [Work supported by ONR.]

10:50

4aUW9. Laboratory measurements of sound propagation in a continuously stratified ocean containing internal waves. Likun Zhang, Harry L. Swinney (Ctr. for Nonlinear Dynam., Univ. of Texas at Austin, 2515 Speedway, Stop C1610, Austin, TX 78712-1199, lzhang@chaos.utexas.edu), and Ying-Tsong Lin (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

A tank experiment is being conducted to explore sound propagation in a continuously stratified ocean containing internal gravity waves that produce fluctuations in acoustic signals. The tank is filled with water of increasing salinity with increasing depth in the fluid; the variation in salinity results in a 150 m/s variation of the sound speed over the 50 cm-depth fluid. Sound arrival times determined in three-dimensional hydrophone scans map out the sound diffraction pattern in the stratified fluid. Good agreement is found between the measurements and results obtained from wavefront modeling by ray acoustics. Sound fluctuations produced by internal waves are measured for several acoustic track locations and at different phases of the internal waves that radiate from a mechanical wave generator. Internal wave fields measured by particle image velocimetry provide input for a parabolic equation model, which is used to predict the sound fluctuations for comparison with measurements. This research is designed to improve the understanding and modeling of sound propagation and scattering by internal waves in the realistic oceans. [Research support: The 2013-14 ASA F. V. Hunt Postdoctoral Research Fellowship (L. Zhang) and ONR MURI Grant N000141110701 (WHOI).]

Contributed Papers

11:10

4aUW10. Comparison of measured acoustic reflection fluctuations and estimates based on roughness. Nicholas P. Chotiros and Marcia J. Isakson (Appl. Res. Labs., Univ. of Texas at Austin, PO Box 8029, Austin, TX 8713-8029, chotiros@arlut.utexas.edu)

The extent to which the roughness of the seabed can account for the acoustic reflection fluctuations is investigated. The roughness of selected areas of the seabed off Panama City, FL, was measured with a laser profiler, as part of Target and Reverberation EXperiment of 2013 (TREX13). The site may be characterized as having small-scale roughness due to bioturbation overlying larger sand ripples due to current activity. It was largely composed of sand with shell hash crossed by ribbons of softer sediment at regular intervals. Simultaneous with the roughness measurement, the seabed reflection was measured with a wide beam echo sounder centered about 10 kHz. Both instruments were deployed on a remotely operated vehicle (ROV) at a height of approximately 2 m above the seabed. Significant fluctuations in the acoustic reflection were observed. By modeling the acoustic scattering due to the roughness, the extent to which the roughness can account for the acoustic fluctuations will be investigated. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

11:25

4aUW11. Influence of short time scale water column fluctuations on broadband signal intensity and beam spreading. Justin Eickmeier and Mohsen Badiey (College of Earth, Ocean and Environment, Univ. of Delaware, Robinson Hall Rm. 112B, Newark, DE 19716, jeickmei@udel.edu)

The environment in a recent experiment exhibited short time scale isotherm depressions and elevations in the temperature profile of the water column. This dynamic behavior is significantly pronounced over a 2 hour period (between 70–90 m depth) during a 24 hour deployment. High frequency broadband transmissions (22–28 kHz) were sent between a stationary source (5 m above the seafloor) and an 8-element vertical hydrophone array (4.5 m above the seafloor) in an approximate depth of 100 m with 1 km separation. Vertical beamforming of measured impulse response across all array elements and application of Gaussian steering revealed strong correlation between vertical temperature profiles and angular spread of the direct path receptions. Inherently a 3D problem, we consider a 2D approach to show beam fluctuations as a function of the environment. 2D PE modeling is driven by measured sound speed profiles to calculate the acoustic field between source and receiver and to beamform across an ideal vertical array for data/model comparison. Over time, fluctuations in the intensity of the acoustic beam, spatial path and angular spread of the direct path signal can be attributed to the vertical oscillations of isotherms in the water column. [Work supported by ONR321.]

11:40

4aUW12. An assessment of the effective density fluid model for scattering from poroelastic sediments with inclusions. Anthony L. Bonomo, Nicholas P. Chotiros, and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

The effective density fluid model (EDFM) was developed to approximate the behavior of sediments governed by Biot's theory of poroelasticity. Previously, it has been shown that the EDFM predicts reflection coefficients and backscattering strengths that are in close agreement with those of the full Biot model for the case of a homogeneous poroelastic half-space. However, it has not yet been determined to what extent the EDFM can be used in place of the full Biot model for other cases. In this work, the finite element method is used to compare the scattered pressure predicted using the EDFM with the predictions of the full Biot model for the case of a poroelastic half-space with a single inclusion. The geometry is assumed axisymmetric and the inclusions are allowed to be either a fluid or an elastic solid. [Work supported by ONR, Ocean Acoustics.]

Session 4pAA

Architectural Acoustics: Classroom and Other Room Acoustics

Richard J. Ruhala, Chair
Mechanical Engineering, Kennesaw State University, 1100 South Marietta Parkway, KSU - SPSU - ME Department, Marietta, GA 30060

Contributed Papers

1:15

4pAA1. Acoustical design of schools on military bases: Meeting the needs of DoDEA's 21st century school Initiative and LEED acoustical requirements. Jessica S. Clements (Newcomb & Boyd, 303 Peachtree Ctr. Ave. NE, Ste. 525, Atlanta, GA 30303, jclements@newcomb-boyd.com)

In their efforts to improve student learning environments, DoDEA (Department of Defense Education Activities) has included acoustical requirements in their 21st Century Schools initiative design guidelines. Recently, DoDEA projects have been required to also achieve LEED certification. The acoustical requirements for LEED schools outlined in IEQ prerequisite 3 (IEQp3) Minimum Acoustical Performance are referenced in part from ANSI S12.60 Acoustic Performance and Guidelines for Schools. Ideally the schools would achieve both the LEED acoustic prerequisite and the additional acoustic credit in order to achieve most of the goals of ANSI S12.60. However, meeting some aspects of the 21st Century Schools design guideline seems to preclude achieving the additional Acoustical Performance credit. Each of these guidelines includes different aspects of architectural acoustics design requirements. This leads to potentially three different design guidelines affecting a single project. The presentation will compare and contrast the design requirements of ANSI S12.60-2010 Part 1, LEEDv4 Minimum Acoustic Performance IEQp3, LEEDv4 Acoustic Performance Credit IEQc9, and the DoDEA 21st Century Design Guideline. Discussion will include examples from recent projects and resolutions used between conflicting design requirements.

1:30

4pAA2. An investigation into the acoustics of different sized open plan and enclosed Kindergarten classrooms. Kiri T. Mealings (Linguist, Macquarie Univ., Level 3 AHH Macquarie University, Sydney, New South Wales 2936, Australia, kiri.mealings@students.mq.edu.au), Jörg M. Buchholz (National Acoust. Labs., Sydney, New South Wales, Australia), Katherine Demuth (Linguist, Macquarie Univ., Sydney, New South Wales, Australia), and Harvey Dillon (National Acoust. Labs., Sydney, New South Wales, Australia)

Open plan classrooms, where several class bases share the same space, have recently re-emerged in Australian primary schools. This study compared the acoustics of four different Kindergarten classrooms: an enclosed classroom with 25 students, a double classroom with 44 students, a linear fully open plan triple classroom with 91 students, and a semi-open plan K-6 classroom with 205 students. Ambient noise levels, intrusive noise levels, occupied background noise levels, and teacher's speech levels were recorded during different activities. Room impulse responses using logarithmic sweeps were also recorded for different teaching scenarios. From these recordings, signal-to-noise ratios, speech transmission index scores, and reverberation times were calculated. The results revealed much higher intrusive noise levels in the two largest open plan classrooms, resulting in signal-to-noise ratios and speech transmission index scores to be well below those recommended in classrooms with students of this age. Additionally, occupied background noise levels in all classrooms were well above recommended levels. These results suggest students may have difficulty listening and learning in open plan classrooms and teachers are likely to strain their voice. Therefore, open plan classrooms are unlikely to be appropriate learning environments for young children unless high intrusive noise levels are moderated.

1:45

4pAA3. Sound levels and speech privacy in an open space academic learning environment. Richard J. Ruhala, Aaron Baker, and Ermal Shpuza (Kennesaw State Univ., 1100 South Marietta Parkway, KSU - SPSU - ME Dept., Marietta, GA 30060, rruhala@spsu.edu)

Undesired sound and speech privacy in a large educational space or open office environment are persistent problems for occupants. The architectural design studio on the campus of Kennesaw State University (formerly Southern Polytechnic State University) is used as a case study. Students have desk space very similar to those in professional open offices, and are used for individual and group projects. Simultaneously classes are held where instructors or students will be presenting to their class in an area that is only partially partitioned off from the rest of the design studio. For the first set of *in situ* experiments, sound levels are recorded over several days at various locations and correlated with learning activity. Second, broadband noise is used to measure the reduction of spectrum levels between source and receiver, at different locations. Finally, students and professors participate in an anonymous questionnaire regarding their perceptions of the acoustic quality of the space; specifically regarding the speech intelligibility of professors, level of noise distractions, and preferences of when to work in the studio. This study aims to understand the acoustic impact of human activity occurring concurrently in a large open academic environment.

2:00

4pAA4. Classroom acoustics for vocal health of elementary school teachers. Jennifer K. Whiting, Zachary R. Jensen, Mark L. Berardi, Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, lundjenny@comcast.net), and Eric J. Hunter (Dept. of Commun. Sci. and Disord., Michigan State Univ., East Lansing, MI)

School teachers have an elevated risk of voice problems due to vocal demands in the workplace. ANSI SI2.60-2002 provides a standard for classroom acoustics, but it focuses primarily on students and unoccupied classroom settings. This presentation explores a preliminary study of six elementary school teachers that included measurements of architectural acoustics parameters and noise-levels of their classrooms, as well as their speech levels and fundamental frequencies over the course of a school day. The measurement methods and speech trends are discussed for the various cases, demonstrating that classroom acoustics standards may benefit from greater attention to teacher vocal health.

2:15

4pAA5. Identifying vocal changes due to environmental changes: Real-time Lombard effect. Zachary R. Jensen, Jennifer K. Whiting, Timothy W. Leishman (Acoust. Res. Group, Phys. and Astronomy Dept., Brigham Young Univ., Provo, UT 84602, zjens1@gmail.com), and Eric J. Hunter (Commun. Sci. & Disord., Michigan State Univ., East Lansing, MI)

School teachers are known to have an elevated risk of voice problems due to speaking demands in the workplace, as well as noise conditions in the classroom. Six elementary school teachers were recorded for a full school day using a neck-worn accelerometer and microphone. Classroom noise levels were also recorded using seven stationary microphones distributed randomly throughout the room. A method was developed that used the accelerometer signal to identify when a teacher is speaking, subsequently calculating a speech-to-noise ratio from six hours of teaching. Speech levels and fundamental frequencies were assessed through the accelerometer signal. The former were compared to temporally and spatially proximate noise levels from the neck-worn microphone when the teacher was not speaking. The speech data was also compared to spatially averaged and temporally proximate classroom noise levels when the teacher was not speaking. While unique in the duration of the observation for the various teachers, the speech-to-noise ratio exhibited trends related to those suggested in previous studies and is congruent with the Lombard effect. A good understanding of how a teacher compensates for realistic occupational noise conditions can lead to better recommendations for the teacher to retain good vocal health. The method and speech trends will be discussed, and future methodological improvements will be suggested.

2:30

4pAA6. Beyond Lombard: Quantitative changes due to noise and reverberation. Nathan G. Eyring, Mark L. Berardi, Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., N-283 ESC,Provo, UT 84602, eyring.acoustics@gmail.com), and Eric J. Hunter (Dept. of Commun. Sci. and Disord., Michigan State Univ., East Lansing, MI)

Forty-five subjects performed a short vocal task in two different rooms: a variable-acoustics room and an anechoic chamber. The subjects were taken back and forth between the two rooms using a deception protocol. Each time they entered the variable-acoustics room, its acoustical characteristics had been changed without a change in visual appearance. The changing characteristics involved two background noise conditions and two reverberation conditions. Subjects responded to questions about their comfort and perception of the environmental changes. Analysis was performed on the second and third sentences of the rainbow passage. Objective acoustical metrics and perceptual responses were compared for the different settings. In contrast to males, females raised their fundamental frequency (F0) in concert with their raised vocal levels in response to changes in both loudness and reverberation. A high correlation existed between pitch strength and F0. Factor analysis also revealed that F0 and vocal level were more correlated in females than males.

2:45

4pAA7. Acuity to noise and occupational voice risks in simulated acoustical environments. Mark L. Berardi, Jennifer K. Whiting, Nathan G. Eyring, Michael K. Rollins, Zachary R. Jensen (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ.,N283 ESC, Provo, UT 84602, markberardi12@gmail.com), Eric J. Hunter (Dept. of Commun. Sci. and Disord., Michigan State Univ., East Lansing, Utah), and Timothy W. Leishman (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

School teachers are known to have an elevated risk of voice problems due to the vocal demands in their work environments. Forty-five participants (20 females, 25 males, 7 elementary school teachers, and 38 college-age adults) performed a short vocal task in two different rooms: a variable-acoustics room and an anechoic chamber. The subjects were taken back and forth between the two rooms using a deception protocol. Each time they entered the variable-acoustics room, the acoustical characteristics (two background noise conditions and two reverberation conditions) had been changed without a visual appearance of change. Analysis was conducted on recorded second and third sentences of the first paragraph of the Rainbow Passage. Results revealed that differences in response to reverberation was gender specific. Additionally, school teachers seemed to be more susceptible to the noise condition.

3:00–3:15 Break

3:15

4pAA8. Low-cost redesign of a music rehearsal space for high school ensembles. Gabe Murray (Mech. Eng. Dept., Iowa State Univ., 2025 Black Eng., Ames, IA 50011, glmurray@iastate.edu) and Thomas Burns (Starkey Hearing Technologies, Eden Prairie, MN)

This presentation describes the redesign of an existing multi-purpose practice room used by high school concert bands, drumlines, and choral groups. Resource and cost restrictions limited the redesign and evaluation process for surface finishes only. The existing space had minimal early reflections and reverberation due to copious amounts of thick acoustic wall panels, resulting in a dead sound for choral and band use. The biggest challenge was to reposition the wall panels by distributing them in a checkerboard pattern, thereby balancing them with the reflective surfaces of the painted cinderblock underneath. In addition, one third of the absorptive suspended ceiling tile was replaced with 5/8" gypsum panels, and one third of the ceiling tile was removed altogether in order to couple the interstitial space above the suspended ceiling and increase the effective volume of the space. Odeon was used to create auralizations of various configurations by convolving anechoic recordings of a snare drum with the computed impulse responses, which will be demonstrated. This work was done as part of the Eagan, MN, High School Mentor Program and a professional partnership with Starkey Hearing Technologies.

3:30

4pAA9. The perceptual influence of the cabin acoustics on the reproduced sound of a car audio system. Neofytos Kaplanis, Søren Bech (Bang & Olufsen, Peter Bangs Vej, 15, Struer 7600, Denmark, neo@bang-olufsen.dk), Sakari Tervo, Jukka Pätynen, Tapio Lokki (Dept. of Media Technol., Aalto Univ., Espo, Finland), Toon v. Waterschoot (Elec. Eng., ESAT-STADIUS/ETC, KU Leuven, Leuven, Belgium), and Søren H. Jensen (Electron. Systems, Aalborg Univ., Aalborg, Denmark)

A significant element of audio evaluation experiments is the availability of verbal descriptors that can accurately characterize the perceived auditory events. In terms of room acoustics, understanding the perceptual effects of the physical properties of the space would enable a better understanding of its acoustical qualities, and stipulate perceptually relevant ways to compensate for the subsequent degradations. In contrast to concert halls, perceptual evaluation of everyday-sized and less reverberant spaces has been a challenging task, and literature on the subject is limited. In this study, a sensory evaluation methodology [Lokki *et al.*, J. Acoust. Soc. Am. **132,** 3148–2161 (2012)] was employed to identify the most relevant attributes that characterize the influence of the physical properties of a car cabin on the reproduced sound field. A series of *in-situ* measurements of a high-end car audio system was performed for different physical settings of the car's cabin. A novel spatial auralization methodology was then used, and participants were asked to describe verbally the perceived acoustical characteristics of the stimuli. The elicited attributes were then analyzed following a previous review [Kaplanis *et al.*, in *55th Int. Conf. Aud. Eng. Soc.* (2014)] and possible links to the acoustical properties of the car cabin are discussed. [This study is a part of Marie Curie Network on Dereverberation and Reverberation of Audio, Music, and Speech. EU-FP7 under agreement ITN-GA-2012-316969.]

3:45

4pAA10. Investigations on multi-slope energy decay formation within single-space monumental structures by diffusion equation modeling and intensity probe measurements. Zühre Sü Gül (MEZZO Studyo LTD, METU/ODTU Technopolis Galyum Blok No 21 - A, Cankaya, Ankara 06800, Turkey, zuhre@mezzostudyo.com), Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY), and Mehmet Čalışkan (Mech. Eng., Middle East Tech. Univ., Ankara, Turkey)

Multiple domed superstructures with different volumetric and material attributes are previously investigated using in-situ acoustical measurements

4p THU. PM

and acoustical simulation methods for analyzing their contributions to multi-slope energy decay formation. Relevant acoustical parameters including decay rates and decay times are computed by applying the Bayesian decay analysis. Initial results indicate double- or triple-slope decay characteristics for specific measurement configurations for selected cases. The ongoing research is aimed at explaining the probable reasons of multiple decay formation as observed in such single space enclosures. Diffusion equation modeling (DEM) and intensity probe measurements are utilized to observe spatial sound energy distributions and energy flow vectors. Both computed and measured flow vectors indicate the contribution of central dome to the later energy feedback. For a specific case the trial by DEM over virtual model with floor with marble instead of carpet resulted in a centrally concentrated energy, as a result preventing the multi-slope formation. The results support the argument that non-diffuse sound fields due to geometrical and material characteristics of superstructures provide the circumstances for non-exponential energy decay formation.

4:00

4pAA11. Minimum phase acoustic systems: An intuitive model study. Sahar Hashemgeloogerdi and Mark F. Bocko (Elec. and Comput. Eng., Univ. of Rochester, 405 Comput. Studies Bldg., Rochester, NY 14627, shashemg@UR.Rochester.edu)

The zeros and poles of minimum phase systems lie within the unit circle in the z-plane, which ensures the existence of a stable inverse. In acoustics, the question if a system is minimum phase has important consequences, such as the ability to find an inverse of a room's acoustic transfer function to compensate for reverberation. In this paper, we provide an intuitive model of minimum phase acoustic systems based upon physical arguments. Using the image method, we compute the transfer function of generic one, two, and three-dimensional acoustic enclosures with a sound source and a microphone placed at arbitrary locations. We find the relationship between the direct sound field and the reverberant sound field that ensures the minimum phase property of the system. Essentially, the system retains the minimum phase property if the radius of reverberation, i.e., the distance from the source where the direct sound and reverberant sound become equal in magnitude, lies outside of the boundaries of acoustic enclosure. We show that a one-dimensional acoustic system always is minimum phase for the entire range of source-microphone locations and wall reflection coefficients. However, two and three-dimensional enclosures display minimum phase response for limited ranges of the system parameters.

4:15

4pAA12. Evaluation of a three-way omnidirectional sound source for room impulse response measurements. David A. Dick, Matthew T. Neal, Carol S. Tadros, and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, dad325@psu.edu)

Commercially available omnidirectional loudspeakers, which are used to measure room impulse responses, typically have a limited bandwidth and are only omnidirectional below approximately 1kHz. To extend the operating bandwidth, a three-way omnidirectional source was built consisting of a subwoofer, a mid-frequency dodecahedron, and a high-frequency dodecahedron. The subwoofer contains two 10 in. drivers in a sealed box. The mid-frequency dodecahedron uses 12 4 in. mid-bass drivers. The high frequency dodecahedron was made with 12 closely spaced 3/4 in. dome tweeters, providing omnidirectional behavior up to approximately 5 kHz. The directivities of the dodecahedrons were validated by taking measurements in an anechoic chamber. The performance of the sources in a realistic setting was assessed using measurements taken in Eisenhower Auditorium, located at Penn State. Measurements were made with the three sources placed individually on the stage in the same location and compared to measurements made with the sources stacked one on top of the other. The stacked configuration yielded significantly different results than the individual configuration. Additionally, measurements were made rotating the individual sources on the stage to evaluate the effect of angular orientation on measured parameters. Parameter differences were found to be small up to the 4 kHz octave band. [Work supported by NSF Grant 1302741.]

4:30

4pAA13. St. Bartholomew's parish hall acoustic analysis. Christopher L. Barnobi (CSTI Acoust., 16155 Park Row Ste. 150, Ste. 101, Houston, TX 77084, chris@cstiacoustics.com), Ana Jaramillo (Olson Sound Design, Medellin, Colombia), and Adam Young (CSTI Acoust., Houston, TX)

St. Bartholomew's Catholic Church in Katy, Texas, includes a fellowship hall with a great dome. Parishioners use the space for social events including meals and regular bingo games. High reverberation times in the fellowship hall reduced speech communication and acoustic comfort. Measurements were conducted to assess the reverberation time of the room before and after applying treatment to the space. Sabine reverberation time analysis was used to specify the recommended sound absorption for the fellowship hall. EASE analysis of the space reveals more interesting characteristics.

4:45

4pAA14. What is the necessary level of detail for an acoustic model? Bruce C. Olson and Ana M. Jaramillo (Olson Sound Design, 8717 Humboldt Ave. N, Brooklyn Park, MN 55444, ana.jaramillo@afmg.eu)

In order to evaluate the level of architectural detail needed in an acoustic model, several projects were selected. Using the software EASE, the authors created several models for each project and found that a high level of architectural detail was unnecessary to some extent in an acoustic model, which gained an advantage in modeling and calculation times.

Session 4pAB

Animal Bioacoustics: Audio Playback in Animal Bioacoustics

Benjamin N. Taft, Chair
Landmark Acoustics LLC, 1301 Cleveland Ave., Racine, WI 53405

Chair's Introduction—2:00

Invited Paper

2:05

4pAB1. Multimodal communication in wolf spiders: Playback studies with visual and vibratory signals. George W. Uetz (Biological Sci., Univ. of Cincinnati, ML 0006, Cincinnati, OH 45221-0006, george.uetz@uc.edu), David L. Clark (Biology, Alma College, Alma, MI), Brent Stoffer, Elizabeth C. Kozak, Madeline Lallo (Biological Sci., Univ. of Cincinnati, Cincinnati, OH), and Heather Kane (Biology, Alma College, Alma, MI)

Previous studies of *Schizocosa ocreata* wolf spiders have shown that visual and vibration signals are equally capable of eliciting female receptivity, but that multimodal cues enhance female response. Studies also indicate that signals in both modes convey male quality information; females choose males with larger foreleg tufts or greater amplitude vibratory signals. Male spiders eavesdrop on both the visual and vibration signals of other males and exploit them to find mates. We examined female mate preferences and male eavesdropping using video-vibratory playback experiments. Female *S. ocreata* respond to playback of male courtship, showing responses to both visual and vibratory signals alone, as well as multimodal playback. Females also showed preference for male quality indicators in both sensory modes. Eavesdropping males responded to playback of male courtship with displays of their own, but responded to signal modes differently. Males displayed a higher rate of courtship tapping in response to isolated vibration signals compared to visual or multimodal stimuli. However, in choice tests, males responded with higher display rates with multimodal and visual stimuli. Results suggest that visual, vibratory and multimodal courtship signals provide information about potential mates and rivals, and that responses of males and females may differ depending on context.

Contributed Paper

2:25

4pAB2. Quantifying response in vocal behavior of fin whales to local shipping in the Southern California. John E. Joseph and Tetyana Margolina (Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jejoseph@nps.edu)

Analysis of a year-long passive acoustic dataset collected at Thirtymile Bank from December 2012 to December 2013 revealed two main types of response, if any, in the vocalization behavior of fin whales to passing ships. The whales ceased calling or reduced the vocalization intensity and then resumed it after the ship passes, or the vocalization intensity increased during the passage and remained elevated for a short period of time afterward. While the response has been primarily observed in 20 Hz calls, several cases have been observed when the call pattern changed with 40 Hz calls recorded after the ship passed. Our study quantifies observed response in terms of both vocalization intensity and call rates. The latter is made possible by applying an image-based detection algorithm allowing for extraction of individual calls. Intensity variations and call rate changes related to ship sounds are statistically estimated, as well as the type and strength of the response to ship sound amplitude, frequency content and relative position of the whales to the passing ships. The received level of the ship sounds presumably heard by whales is modeled using an acoustic propagation model and a Monte Carlo approach to estimate the unknown whale location.

Invited Paper

2:40

4pAB3. The potential for acoustic communication in the "purring" wolf spider, *Gladicosa gulosa*. Alexander L. Sweger and George W. Uetz (Biological Sci., Univ. of Cincinnati, P.O. Box 210006, Cincinnati, OH 45221, swegera@gmail.com)

Vibration is an important part of the sensory world in spiders, and many species have adapted vibration as a major part of their conspecific communication. While nearly all male wolf spiders produce vibrations during courtship, the "purring" wolf spider, *Gladicosa gulosa,* also produces an acoustic signal in conjunction with its vibratory display. However, with limited previous research on this species, the evolutionary significance of this component remains unknown. Given that spiders are not known to possess sensory structures

for directly perceiving airborne sound, this raises a number of questions about the production, reception, and possible role of the signal. We measured male signal production and male/female responses to isolated acoustic signals on both vibrating (paper) and non-vibrating (granite) substrates. We found that male signals, both vibratory and acoustic, are only present in vibrating substrates. We also found significant differences in phonotaxis based on sex of the focal individual, stimulus type, and substrate type. These results suggest that the substrate plays an important role in both production and reception of the acoustic signal, and that under certain conditions, acoustic signaling may have a role in the communication network in this species.

Contributed Paper

3:00

4pAB4. Playback experiments for *Thryothorus ludovicianus* in urban backyard experiments. David P. Knobles (KSA, PO Box 8029, Austin, TX 78713, knobles@arlut.utexas.edu) and Mohsen Badiey (Univ. of Delaware, Newark, DE)

Acoustic playback experiments were performed from June 2014 to April 2015 for *Thryothorus ludovicianus* (Carolina Wren) in a backyard environment in Austin, Texas. Acoustic recordings were made for pre-playback, playback, and post-playback periods for multiple song types. The songs were obtained from recordings made in the backyard and surrounding neighborhood. Recordings of both the playback and acoustic responses of the bird subjects are made in coincidence with visual observations of the bird's behavior. Bird behavior is quantified in terms of the statistics of closest distance of approach, latency to approach, latency to sing, number of post-playback songs, number of flybys, percent of male-female duets, temporal correlation of individual song phrases, and number of singing competitions with neighboring subjects. Statistics of the properties of the measured spectrograms and spatial correlation properties for experiments are presented, and attempts are made to interpret the observed behavior and acoustical responses of the birds in the form of communication and territorial defense with the bird's ability to estimate distance using the degradation (transmission loss and reverberation) of the playback calls. It was also observed that *Thryothorus ludovicianus* will respond to playback calls of *Cardinalis cardinalis* (Northern Cardinal) and this interaction between bird species is examined.

Invited Paper

3:15

4pAB5. The use of acoustic playbacks with captive cetaceans. Jason N. Bruck (Dept. of Integrative Biology, Oklahoma State Univ., 501 Life Sci. Way, Stillwater, OK 74075, jbruck@okstate.edu)

Playbacks are a powerful tool in studies of animal communication and cognition. They can provide insights into referentiality, social memory, anti-predator strategies, sexual selection, as well as a host of other topics. In particular, playbacks have been a useful method for understanding the nature of wild dolphin communication using temporarily restrained animals. Unfortunately, there are a series of questions where complete social histories of subjects or certain controls are needed, but unavailable in wild study populations. Captive, free-swimming animals present a complementary set of subjects where one can address questions involving referentiality or dolphin social memory. Methods for conducting playbacks with captive groups differ greatly than for wild subjects. I will explore the procedural differences of these experiments, including overcoming difficulties associated with recording and presenting sounds in an acoustically challenging environment as well as how to measure response behavior. Furthermore, I will discuss the potential of captive acoustic playbacks by highlighting data collected from such studies including evidence for dolphin multi-decade social memory, context dependent cue usage and acoustic kin recognition.

Session 4pBAa

Biomedical Acoustics: Ultrasound Contrast Agents: Molecular Imaging Applications

Tyrone M. Porter, Cochair
Boston University, 110 Cummington Mall, Boston, MA 02215

Jonathan A. Kopechek, Cochair
University of Pittsburgh, 200 Lothrop Street, Pittsburgh, PA 15213

Invited Papers

1:00

4pBAa1. Molecular imaging using ultrasound and microbubbles. Flordeliza S. Villanueva (Medicine, Univ. of Pittsburgh, 200 Lothrop St., Pittsburgh, PA 15213, villanuevafs@upmc.edu)

Increasing insight into the molecular mechanisms of disease has created a need to visualize sub-cellular events in clinical populations in order for treatment paradigms to advance. Molecular imaging has emerged as an important translational tool for understanding disease pathogenesis, for which various probes and detection methods have been developed. Ultrasonic detection of molecular markers of disease using acoustically active particles is one such *in vivo* modality. Molecular imaging with ultrasound utilizes intravascular delivery of gas-filled microspheres (microbubbles) targeted to bind to disease-specific epitopes expressed by vascular endothelial cells. Microbubble binding occurs by virtue of targeting ligands attached to the microbubble surface, which confer specific binding to the endothelial target. In the presence of ultrasound, the bound microbubbles emit an acoustic signal which can be detected as a persistent contrast effect during ultrasound imaging with non-linear based detection strategies. In pre-clinical models, ultrasound molecular imaging has been used to detect early atherosclerotic disease, myocardial ischemic memory (recent myocardial ischemia), acute heart transplant rejection, angiogenesis, and for *in vivo* tracking of therapeutically administered stem cells. Such applications raise exciting possibilities for meaningful bedside translation of this imaging platform to address important clinical issues in an era of molecular medicine.

1:30

4pBAa2. Novel polymer-lipid assemblies for stimulus-responsive imaging contrast agents. Andrew P. Goodwin (Chemical and Biological Eng., Univ. of Colorado Boulder, 3415 Colorado Ave., 596 UCB, Boulder, CO 80303, andrew.goodwin@colorado.edu)

Stimulus-responsive macromolecular structures are of tremendous importance for the next generation of tools for *in vivo* imaging and site-directed therapy. In this talk, I will present our research group's work regarding the creation of self-assembled amphiphilic structures that respond to the presence of biomarkers that present as hallmarks for disease such as deep venous thrombosis and cancer. In one instance, the mechanical properties of synthetic, lipid-shelled microbubbles could be manipulated to sense for the biomarker thrombin using DNA hybridization. These microbubbles were found to have "on-off" ratios of ~100 and could highlight regions of elevated thrombin in real time in a live rabbit model. In another research direction, low-contrast nanoemulsions were designed to vastly increase their detection capabilities in response to chemically-induced changes in their surface properties. Both examples showcase a method for obtaining site- or biomarker-specific contrast enhancement for deep tissue imaging using commercially available technologies and have excellent promise for advancement to clinical use.

2:00

4pBAa3. Molecular imaging with ultrasound: Steps toward clinical translation. Juergen K. Willmann (Radiology, Stanford, 300, Pasteur Dr., Rm. H1307, Stanford, CA 94305-5621, willmann@stanford.edu)

Molecular imaging provides the ability to non-invasively assess expression levels of molecules by measuring imaging signals generated with the help of molecularly targeted contrast agents. These contrast agents can be directed to bind various molecular targets, thereby visualizing and quantifying disease processes at the molecular level *in vivo*. In recent years, the field of molecular imaging has been rapidly expanding and involves multiple imaging modalities. Among those, ultrasound imaging is a widely available, relatively inexpensive, and real-time imaging technique without exposing patients to radiation. It is already the first-line radiological imaging modality for many organs and disease processes in patients. By combining the these advantages of ultrasound with the ability to image molecular signatures using novel molecularly targeted ultrasound contrast agents, molecular ultrasound imaging is a quickly evolving imaging strategy that has great potential as a highly accurate and quantitative method for various clinical applications, including earlier detection of for example cancer, characterization of focal lesions, and quantitative monitoring of various disease processes at the molecular level. The objective of this presentation is to provide an overview on the principles of ultrasound molecular imaging and to highlight current translational studies to move ultrasound molecular imaging approaches into first in human clinical applications.

2:30

4pBAa4. Laser-induced vaporization and acoustic-induced cavitation of droplets. Jaesok Yu (Dept. of BioEng., Univ. of Pittsburgh, 3550 Terrace St., Scaife Hall 958, Pittsburgh, PA 15261, JAY49@pitt.edu), Man M. Nguyen, and Kang Kim (Dept. of Medicine, Univ. of Pittsburgh, Pittsburgh, PA)

In this study, micro-droplets developed for photoacoustic imaging and drug delivery have been evaluated *in vitro*. Induced by a short laser pulse, perfluoropentane mixed with optical dye was vaporized and expanded up to about 20 times of initial diameter of 3–10 micron, generating strong broadband photoacoustic signals. It was found these vaporized droplets became less stable and prone to rupture to ultrasound pulses. The broadband inertial cavitation signals by a very short ultrasound pulse can provide high spatial resolution in passive cavitation imaging. For feasibility test, passive cavitation imaging algorithms were implemented in a commercial ultrasound open platform with a linear array transducer, centered at 5 MHz. The system was synchronized with a 1 MHz unfocused single element transducer. 700kPa single-cycle excitation ultrasound pulse was induced to a 580 μm inner diameter polyethylene tube containing micro-droplets. The cavitation imaging was performed before and after vaporizing droplets by laser. Broadband emissions of 3–7MHz were observed only with vaporized droplets. These preliminary results show the feasibility of cavitation imaging of vaporized droplets with a short ultrasound excitation pulse for improved spatial resolution and could lead to further *in-vivo experiments*.

2:45

4pBAa5. Numerical investigation of the nonlinear attenuation and dissipation of acoustic waves in a medium containing ultrasound contrast agents. Amin Jafari Sojahrood, Raffi Karshafian, and Michael C. Kolios (Dept. of Phys., Ryerson Univ., 350 Victoria St., Toronto, Ontario M5B2K3, Canada, amin.jafarisojahrood@ryerson.ca)

The presence of microbubble ultrasound contrast agents (UCAs) in path of an ultrasound beam increases the attenuation of the medium. A detailed knowledge about the medium attenuation in the presence of UCAs is critical for optimizing ultrasound applications. In this study, a model incorporating the nonlinear attenuation of microbubbles is developed by deriving the complex and real part of the wave number from the Calfish model. Results showed that when UCAs are sonicated by their linear resonance frequency (fr), the effective attenuation of the medium can potentially decrease as the pressure increases, in agreement with experimental observations of the attenuation of monodisperse microbubbles. When sonicated with their pressure dependent resonance frequency (fs), the effective attenuation of the medium is smaller than the case of sonication with fr, but only below a pressure threshold (Ps). Above Ps, the effective attenuation increases abruptly (e.g. ~3 fold) and becomes significantly more than the attenuation during sonication with fr. The attenuation in the subharmonic (SH) regimes of oscillations (sonication with n*fr where n = 2, 3.) are considerably smaller compared to the cases of sonication with fr and fs (~10 times). The attenuation undergoes a sharp increase concomitant with the generation of SH oscillations.

THURSDAY AFTERNOON, 21 MAY 2015 KINGS 2, 3:15 P.M. TO 5:30 P.M.

Session 4pBAb

Biomedical Acoustics: Therapeutic Ultrasound and Bioeffects

Parag V. Chitnis, Chair
Department of Bioengineering, George Mason University, 4400 University Drive, 1G5, Fairfax, VA 22032

Contributed Papers

3:15

4pBAb1. An equivalent source model for simulating high intensity focused ultrasound fields using a nonlinear parabolic equation. Vera A. Khokhlova, Petr V. Yuldashev, Pavel B. Rosnitskiy (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow 119991, Russian Federation, va.khokhlova@gmail.com), Wayne Kreider, Adam D. Maxwell, Michael R. Bailey (Ctr. for Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), and Oleg A. Sapozhnikov (Dept. of Acoust., Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

The nonlinear parabolic Khokhlov-Zabolotskaya (KZ) equation is commonly used for simulation of nonlinear focused fields generated by medical ultrasound transducers. Axially symmetric codes can be easily implemented on personal computers with simulation times on the order of tens of minutes even for the most extensive simulations of shock formation conditions. However, model accuracy can be limited for high intensity focused ultrasound sources with large convergence angles. In previous studies, it was shown that nonlinear parabolic modeling can provide good matching with measurements and full diffraction modeling in the focal region for both single element piston sources and multi-element arrays. Boundary conditions for the modeling were adjusted by varying the aperture of an equivalent planar source with a uniform pressure magnitude and a phase distribution for parabolic focusing. With this approach, there were slight discrepancies in the lateral pressure distributions in the focal plane. Here, it is proposed that an additional variation in the position of the equivalent source enables matching both axial and lateral distributions of the real HIFU sources with higher accuracy. Several examples are presented to compare modeling and measurements of the fields generated by laboratory and clinical HIFU transducers. [Work supported by RSF 14-12-00974 and NIH EB007643.]

3:30

4pBAb2. A sum-of-harmonics time-domain method to distinguish harmonic and broadband signals during passive acoustic mapping of ultrasound therapies. Erasmia Lyka, Christian Coviello (Inst. of Biomedical Eng., Dept. of Eng. Sci., Univ. of Oxford, Old Rd. Campus Res. Bldg., Headington, Oxford OX3 7DQ, United Kingdom, erasmia.lyka@eng.ox.ac.uk), Richard Kozick (Bucknell Univ., Lewisburg, PA), and Constantin C. Coussios (Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Passive Acoustic Mapping (PAM), a novel technique for real-time monitoring of ultrasound-based therapies, performs passive beamforming of non-linear acoustic emissions simultaneously received on an array of transducers. These emissions can be classified according to their frequency content into harmonic, indicative of nonlinear scattering, stable cavitation or tissue boiling, and broadband, indicative of inertial cavitation. However, the magnitude of coherent reflections often greatly exceeds that of incoherent broadband emissions arising from inertial cavitation and the ability to distinguish these two components is, thus, key to successful detection and mapping of inertial cavitation. We propose a novel time-domain-based filter that uses a parametric model in order to estimate a time-varying amplitude harmonic component in the presence of a lower-amplitude broadband signal. Performance evaluation on simulated and experimental data has shown that the model is able to accurately detect a time-varying amplitude harmonic signal for harmonic-to-broadband amplitude ratio as high as 20 dB, even in short data lengths, decreasing the estimation error by at least 80% compared to conventional comb filtering, and it yields passive acoustic maps of better accuracy and spatial resolution. Sum-of-harmonics is expected to become a valuable component of real-time PAM and significantly contribute toward its clinical adoption.

3:45

4pBAb3. Temperature measurement using backscattered ultrasonic power for non-invasive thermometry during HIFU ablatio. Jeremy Chenot and David Melodelima (LabTAU - INSERM U1032, 151 cours Albert Thomas, Lyon 69003, France, David.Melodelima@inserm.fr)

The temperature dependence of ultrasonic tissue parameters has been reported extensively. In this work, the relationship between changes in ultrasound backscattered power and temperature during HIFU treatments was studied. An analytical model was developed based on attenuation, backscattered coefficient, velocity, and thermal expansion to predict temperature changes. *In vitro* and *in vivo* tests were conducted in liver to confirm simulations. HIFU treatments were performed using a focused transducer working at 3 MHz. The radius of curvature and the diameter were both 70 mm. An ultrasound imaging probe working at 7.5 MHz was placed in the center of the HIFU transducer. Long exposure time (120 s) was used to observe smooth temperature increase from 37 to 80 °C. Simulations predicted that backscattered power increased nearly logarithmically with temperature over the range from 37 °C to 80 °C. The model predicted a linear increase 15 dB. A linear increase 12 dB was measured in ultrasound backscattered power during experiments. The tissue temperature increase estimated using backscattered energy correlated well ($r = 0.79$) with temperature measurements performed using thermocouples. This linear relationship between changes in the backscattered energy and actual temperature was observed up to 80 °C. Successful temperature estimation has allowed creating 2D temperature maps during HIFU treatments

4:00

4pBAb4. Phase I-II study of intra-operative high intensity focused ultrasound in 25 patients with colorectal liver metastase. David Melodelima (LabTAU - INSERM U1032, 151 cours Albert Thomas, Lyon 69003, France, David.Melodelima@inserm.fr), Aurelien Dupre, David Perol, Yao Chen, Jeremy Vincenot (Ctr. Leon Berard, Lyon, France), Jean-Yves Chapelon (LabTAU - INSERM U1032, Lyon, France), and Michel Rivoire (Ctr. Leon Berard, Lyon, France)

The objective of this clinical study was to validate the effectiveness, accuracy, tolerance, and safety of a HIFU treatment developed for the treatment of liver metastases in a prospective, phase I-II trial. The transducer has a toroidal shape (diameter: 70 mm, radius of curvature: 70 mm) and was divided into 32 ring-shaped emitters operating at 3 MHz. HIFU was delivered immediately before scheduled hepatectomy. Ablations were performed on healthy tissue within the areas scheduled for resection. First, 30 ablations were carried out in 15 patients. These ablations were all generated within 40 seconds and on average measured 27.5×21.0 mm^2. The phase I study ($n = 6$) showed that use of the HIFU device was feasible and safe and did not damage neighboring tissue. The phase IIa study ($n = 9$) showed that the area of ablation could be precisely targeted on a previously implanted metallic mark. Ablations were achieved with a precision of 1–2 mm. The phase IIb study ($n = 10$) demonstrated ablation of metastases with safety margins, again prior to planned resection. Using electronic focusing the volume of ablation was adjusted to the size of the tumor from 7 cm^3 (40 s of treatment) to 50 cm^3 (6 min of treatment).

4:15

4pBAb5. A simple animal model for cerebral vasculature rupture due to exposure to intense pressure waves. Marjan Nabili, Priyanka Acharya, Yeon Ho Kim, and Matthew R. Myers (Div. of Appl. Mech., Office of Sci. and Eng. Labs.,Ctr. for Devices and Radiological Health, Food and Drug Administration, 10903 New Hampshire Ave., Bldg. 62, Rm. 2233, Silver Spring, MD 20993, marjan.nabili@fda.hhs.gov)

Understanding the threshold of microvasculature rupture in the central nervous system is critical to evaluating the safety of transcranial therapeutic ultrasound procedures, and treatment planning for blast traumatic brain injury. The goal of this study is to determine the threshold for microvasculature rupture in a simple animal model, as a function of the characteristics of the incident pressure-pulse train. An earthworm model was chosen, as a first step in a sequence of increasingly complex models, and because of its readily accessible large vessel. Following anesthetization, the earthworms were sonicated with 3.3 MHz pulse trains from a high-intensity focused ultrasound (HIFU) transducer. A variety of pulse durations, repetition rates, and amplitudes were considered. The pulse duty cycle was kept low (0.0001 to 0.001) to minimize thermal effects. In cases where rupture occurred within 10 min of exposure, the rupture time was recorded. A noticeable threshold for microvascular damage was observed at a peak negative pressure of about 20 MPa. Beyond this pressure, rupture times decreased rapidly with increasing acoustic pressure. The threshold for damage is likely due to the onset of cavitation, though the mechanisms affecting the rupture time require further study.

4:30

4pBAb6. Broadband attenuation measurements of tissue-mimicking phantoms employed for histotripsy. Michael J. Crowe (Div. of Health, Lung, and Vasculature, Dept. of Internal Medicine, Univ. of Cincinnati, 231 Albert Sabin Way, 3940 Cardiovascular Ctr., Cincinnati, OH 45267-0586, crowem2@xavier.edu), Jason L. Raymond (Biomedical Eng. Program, Univ. of Cincinnati, Cincinnati, OH), Christy K. Holland, and Kenneth B. Bader (Div. of Health, Lung, and Vasculature, Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Tissue-mimicking phantoms have previously been developed for assessing the efficacy of therapeutic ultrasound. A new modality of focused ultrasound, histotripsy, relies on the formation of shockwave-induced cavitation to ablate tissue mechanically. The objective of this study was to use a broadband ultrasound method to determine the frequency-dependent attenuation for histotripsy tissue phantoms. An agar tissue phantom was developed with evaporated milk to mimic the acoustomechanical properties of prostate tissue, a key target for histotripsy. The attenuation of this tissue phantom was measured between 2 and 20 MHz using a through-transmission technique with a single pair of PVDF transducers. The power law dependence of the acoustic attenuation spectra was linear with frequency ($R^2 = 0.98$). The attenuation coefficient of the tissue phantom was 0.62 ± 0.02 dB/cm/MHz, which is lower than that reported for *ex vivo* human prostate at 5 MHz. The broadband measurements techniques utilized provide a straightforward metric for the attenuation spectrum of tissue phantoms developed for shocked insonation regimes.

4:45

4pBAb7. Ultrasound causes stem cell differentiation even in the absence of specific biochemical growth factors. Nirali Shah (Mech. Eng., Swinburne Univ. of Technol., PO Box 218, Hawthorn, VIC, Melbourne, Victoria 3122, Australia), Ursula Manuelpillai (Ctr. for Reproduction and Development, Monash Inst. of Medical Res., Melbourne, VIC, Australia), Yosry Morsi, and Richard Manasseh (Mech. Eng., Swinburne Univ. of Technol., Hawthorn, VIC, Melbourne, VIC 3122, Australia, rmanasseh@swin.edu.au)

Experiments are presented on the effect of low-intensity ultrasound (LIUS) on stem cell differentiation. Previous studies had shown that LIUS can increase stem cell proliferation, and that LIUS in combination with biochemical growth factors (GFs) specific to particular cell lineages resulted in stem-cell differentiation into those lineages. However, studies had not been performed with LIUS alone. Amniotic-membrane-derived mesenchymal stem cells (MSCs) were subjected to continuous-wave insonation at 1 MHz and 0.015–0.072 MPa for periods of 10, 20, and 30 min daily for 15 days. Experiments were done both with a negative control (no insonation) and a positive control: a GF known to cause differentiation into smooth muscle cells (SMCs). The extent of differentiation was quantified by immunohistochemical fluorescence. Results showed that LIUS alone caused MSCs to differentiate into SMCs, with statistically significantly greater expression of SMC markers in the 10 and 20 min groups, with the 20 min group showing the highest differentiation. However, the 30 min group was negatively affected by insonation, with lower proliferation and differentiation. Since MSCs can differentiate into many cell lineages, the results imply that LIUS favors smooth muscle cells, with implications for *in-vivo* treatment of vascular disease and tissue engineering.

5:00

4pBAb8. Designing ultrasound fields to control the morphology of engineered microvessel networks. Eric S. Comeau (Dept. of Biomedical Eng., Univ. of Rochester, 201 Robert B. Goergen Hall, Box 270168, Rochester, NY 14627, eric.comeau@rochester.edu), Denise C. Hocking (Pharmacology and Physiol., Univ. of Rochester, Rochester, NY), and Diane Dalecki (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Spatial patterning of endothelial cells using ultrasound standing wave fields (USWF) can promote extensive microvessel network formation throughout the volume of three-dimensional collagen hydrogels. The goal of this study was to identify acoustic exposure parameters that generate specific initial spatial patterns of cells, in order to control resultant microvessel morphology. Endothelial cells were suspended in soluble collagen and exposed to a 1-MHz continuous wave USWF for 15 min during collagen gel polymerization. Samples were exposed to peak USWF pressures of 0, 0.1, 0.2, or 0.3 MPa. Samples were either imaged immediately post-exposure using high frequency ultrasound or cultured for ten days. Analysis of B-mode ultrasound images confirmed differences in initial cell band spacing and cell band density between the four exposure pressures tested. After ten days in culture, USWF-induced cell patterning resulted in three distinct microvessel morphologies. Specifically, 0.1 MPa exposure resulted in capillary-like networks, 0.2 MPa exposure resulted in non-branching vessel structures, and 0.3 MPa exposure resulted in hierarchical branching microvessel networks. Spatial characteristics of initial cell bands were then correlated with resulting microvessel network morphology. Results of these investigations allow for the capability to predictively, reproducibly, and rapidly form microvessel networks of known morphology within 3D engineered hydrogels.

5:15

4pBAb9. A nonlinear derating method for estimating high-intensity focal pressures in tissue. Seyed Ahmad Reza Dibaji (Dept. of Mech. and Mater. Eng., Univ. of Cincinnati, 2600 Clifton Ave., Cincinnati, OH 45221, dibajisa@mail.uc.edu), Yunbo Liu, Joshua E. Soneson (Office of Sci. and Eng. Labs., Ctr. for Devices and Radiological Health, U.S. Food and Drug Administration, Silver Spring, MD), Rupak K. Banerjee (Dept. of Mech. and Mater. Eng., Univ. of Cincinnati, Cincinnati, OH), and Matthew R. Myers (Office of Sci. and Eng. Labs., Ctr. for Devices and Radiological Health, U.S. Food and Drug Administration, Silver Spring, MD)

Methods for converting high-intensity focused ultrasound (HIFU) pressure measurements made in water to values appropriate for tissue have considerable value in preclinical testing and treatment planning. Such "derating" methods are straightforward in the linear-acoustics regime, but are much more difficult at higher powers. In this study, a nonlinear derating method is used to estimate focal pressure in a tissue phantom. The on-axis pressure in water was recorded using a hydrophone. Fourier transformation of the recorded pressures was performed and the resulting modal amplitudes were reduced using a combination of "source scaling" (measurements in water performed at a lower source pressure than in tissue phantom) and "endpoint scaling" (amplitudes reduced at the target location). The reduced modal amplitudes were used in the convolution term of the evolution equation to determine the pressure in tissue. The focal pressure waveform estimated by this method was compared with a direct measurement in a tissue phantom. The results show that, with the proper combination of source and endpoint scaling, focal pressure in a tissue phantom can be reproduced by derating within 15% error.

Session 4pEA

Engineering Acoustics: General Topics in Engineering Acoustics

Roger T. Richards, Chair
US Navy, 169 Payer Ln., Mystic, CT 06355

Contributed Papers

1:00

4pEA1. Aerodynamic and acoustic analysis of an industrial fan. Yi Liu (Ingersoll Rand, 800 Beaty St., Davidson, NC 28027, yi.liu@irco.com), Jeremy Bain (Bain Aero LLC, Atlanta, GA), Gang Wang (Ingersoll Rand, La Crosse, Wisconsin), Mike Lucas (Ingersoll Rand, Davidson, NC), and Percy Wang (Ingersoll Rand, Tyler, Texas)

The efforts to predict noise radiation for an industrial fan using direct computational fluid dynamics (CFD) simulation is presented in this paper. Industry has been using CFD tool to guide fan design in terms of efficiency prediction and improvement. However, the use of CFD tool for aerodynamic noise prediction is very limited in the past, partly due to the fact that research in aero-acoustics field was not practical for industry application. With the most recent technologies in CFD field, the industry application of aero-acoustics becomes much more promising. It is demonstrated here that fan tonal noise and broadband noise at low frequencies can be predicted using Overset grid system and high order finite difference schemes with acceptable fidelity.

1:15

4pEA2. An analysis of multi-year acoustic and energy performance data for bathroom and utility residential ventilation fans. Wongyu Choi, Antonio Gomez (Mech. Eng., Texas A&M Univ., Riverside Energy Efficiency Lab., 3100 State Hwy. 47, Bryan, TX 77807, wongyuchoi@tamu.edu), Michael B. Pate (Mech. Eng., Texas A&M Univ., College Station, TX), and James F. Sweeney (Texas A&M Univ., Bryan, TX)

Loudness levels have been established as a new requirement in residential ventilation standards and codes including ASHRAE and IECC. Despite the extensive application of various standards and codes, the control of loudness has not been a common target in past whole-house ventilation standards and codes. In order to evaluate the appropriate loudness of ventilation fans, especially in terms of leading standards and codes, a statistical analysis is necessary. Therefore, this paper provides statistical data for bathroom and utility ventilation fans over a nine year period from 2005 to 2013. Specifically, this paper presents an evaluation of changes in fan loudness over the 9 year test period and the relevance of loudness to leading standards including HVI and ASHRAE. The loudness levels of brushless DC-motor fans are also evaluated in comparison to the loudness of AC-motor fans. For AC and DC motor fans, relationships between loudness and efficacy was determined and then explained with regression models. Based on observations, this paper introduces a new "loudness-to-energy ratio" coefficient, L/E, which is a measure of the acoustic and energy performance of a fan. Relationships between acoustic and energy performances are established by using L/E coefficients with supporting statistics for bathroom and utility fans.

1:30

4pEA3. Evaluation and validation of the Federal Highway Administration: Traffic noise model in a sanitized environment. Monty A. Rahman (Civil Eng., Morgan State Univ., 8507 Horseshoe Rd., Ellicott City, MD 21043, rahman@mars-group.net)

The development of powerful physics based simulation models that are dependent on stochastic parameters does not always guarantee reliability. The Federal Highway Traffic Noise Model (TNM) is such a complex model that relies on Reference Energy Mean Emission Level (RMEL) from highway vehicle-type average speed and volume to predict noise propagation in far-field environment. Although the TNM accounts for static parameters such as ground type, it does not account for changing noise source or noise propagation medium. Currently, when validating TNM, the neglected dynamic parameters in the model are usually accounted for by calibrating the model to agree with measured data. However, a model is not necessarily validated when its predicted output is compared to measured data and updated to produce desired values. This paper will review current validation procedures and assess TNM's reliability to predict noise propagation from a highway line source within a sanitized environment. Sanitized environment is an open field within which noise propagation is independent of dynamic variables. By validating TNM within a sanitized environment TNM's residual prediction error due only to variation in the traffic parameters (vehicle type, speed, and flow rate) could be identified and the reliability of REML used in TNM could be confirmed.

1:45

4pEA4. An improved two-microphone transfer function method for measuring oblique absorption coefficient in the free field using numerical modeling. Hubert S. Hall (Signatures, Naval Surface Warfare Ctr. Carderock Div., 620 Michigan Ave. NE, Washington, DC 20064, 61hall@cardinalmail.cua.edu), Joseph F. Vignola, John A. Judge, Diego Turo (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC)

The use of the two-microphone free field transfer function technique has remained unchanged since its development. In practice, finite sample sizes limit measurement fidelity due to edge effects. The sound field contribution from edge diffraction has generally restricted the technique from use at low frequencies (<300 Hz) and required test panels greater than several square meters in area. This effort intends to characterize the diffraction contribution to the sound field for relatively small panels. Using numerical modeling, samples of like acoustic properties were excited by a point source at normal incidence to quantify the diffraction term. Following validation via comparison with impedance tube data, the diffraction term was incorporated in an updated derivation of the complex reflection coefficient equation and validated experimentally for a known reference material at both normal and oblique incidence. The goal of this work is to validate measurements of oblique absorption coefficient in an anechoic chamber for panels smaller than one square meter at frequencies down to 100 Hz.

2:00

4pEA5. Initial laboratory experiments to validate a phase and gradient estimator method for the calculation of acoustic intensity. Darren K. Torrie, Eric B. Whiting, Kent L. Gee, and Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., N223 ESC, Provo, UT 84602, darren@torriefamily.org)

A recently developed phase and gradient estimator (PAGE) method for calculating acoustic intensity from multiple pressure measurements [Thomas *et al*., J. Acoust. Soc. Am. **134**, 4058 (2013)] has been tested via anechoic laboratory measurements of the radiation from multiple loudspeakers. The measurements were used to examine the effects of probe geometry, size, frequency range, and source characteristics on the active intensity calculated from both the PAGE and the traditional cross-spectral approaches. Preliminary results are shown for multiple probe and source configurations, and confirm that the PAGE method results in a broader valid frequency range for a given probe geometry.

2:15

4pEA6. Reverberation mapping using instantaneous intensity estimates from a tetrahedron microphone array. Thomas Burns (Starkey Hearing Technologies, 6600 Washington Ave. S, Eden Prairie, MN 55344, tburns@starkey.com)

For hearing aids, the directivity index is a benchmark defined under two acoustical conditions that a hearing-aid user won't encounter, namely, the ratio of sound power signal arriving from the on-axis target in an anechoic condition to the isotropic spherical noise. It would be useful to benchmark the speech signal to noise that is encountered in a typical environment. The purpose of this study is to map the instantaneous acoustic intensity in a room using a head and torso voice simulator as the source and a regular tetrahedron microphone array in the field. Four impulse responses from a small tetrahedron were measured in a reverberant conference room of gypsum walls, carpet, and absorptive ceiling tile. Welch's method was used to compute one-third octave estimates of the auto- and cross-spectra from the impulse responses, and these spectra were used to estimate the steady-state, 3D instantaneous acoustic intensity vector via the time-averaged active intensity and the maximum amplitude of the reactive intensity. A histogram of the instantaneous intensity vector revealed an angular estimate for the arriving sound power that was, in general, cylindrically rather than spherically "diffuse."

2:30

4pEA7. Acoustic characterization of micro-perforate porous plates. Martha C. Brown (AeroAcoust. Branch, NASA Langley Res. Ctr., 1D East Reid St., Rm. 105C, M.S. 164D, Hampton, VA 23681, martha.c.brown@nasa.gov), Michael G. Jones, and Brian M. Howerton (Structural Acoust. Branch, NASA Langley Res. Ctr., Hampton, VA)

The research objective of this paper is to develop an acoustic impedance model for micro-perforate plates. Passive acoustic liners consisting of perforate plates-over-honeycomb structures are a key contributor in the reduction of fan noise propagated through the inlet and aft-fan duct of aircraft engine nacelles. These perforated plates are physically characterized by plate thickness, t, orifice diameter, d, and porosity, σ, and the resultant liners are typically classified as conventional for $t/d \approx 1$ or micro-perforate for $t/d \gg 1$. Micro-perforate plates are becoming more popular because of their acoustic linearity, i.e., insensitivity to sound level. The goal of the current research is to develop models to better understand the acoustic behavior of liners constructed with micro-perforate facesheets. The first step is to model the flow resistivity of micro-perforate plates, computed as the ratio of the static pressure drop across the plate to the mean flow through the orifices. This result is then used to model the acoustic impedance, which is defined as the ratio of the acoustic pressure to the normal component of acoustic particle velocity at the liner surface. These tests are conducted in the NASA Langley Raylometer and Normal Incidence Tube with samples spanning a range of $4\% \leq \sigma \leq 20\%$ and $5.0 \leq t/d \leq 7.14$.

2:45

4pEA8. Experimental tests of protective covers to obtain optimal transmission loss in porous materials. Jin Liu (UTC Carrier Technol., Methods & Components Eng., 107 Woodmancy Ln., Fayetteville, NY 13066, lewjin@yahoo.com) and John Wang (UTC Carrier Technol., Methods & Components Eng., Syracuse, NY)

Porous resistive materials are widely used to increase the wide-band transmission loss and reduce shell vibrations and noise propagation. For industrial applications, porous materials need to be covered to protect them from environmental contaminants. The goal of our experimental studies is to determine suitable covers which are acoustically transparent and offer this needed protection. Expansion volumes were designed for transmission loss tests with two accurate modeled limits related to two rigid boundary conditions. Seven different covers were tested in this study tests. Best results were achieved using thin porous steel woven cloth. Tyvek, Mylar, thick steel woven cloth, and the Nomex were not as effective. The porous materials and covering materials were tested and then retested after being soaked in oil to explore transmission loss that might occur in actual industrial conditions. These independent measurements of the cover materials were used to determine the material property in the FEA models.

3:00

4pEA9. Unconditionally stable time-domain elastic wave simulations by the alternative direction implicit method. Yu Shao, Myoung An, and Shumin Wang (Auburn Univ., 200 Broun Hall, Auburn, AL 36849, wangs@auburn.edu)

The stability of the conventional staggered-grid finite-difference time-domain (FDTD) method for elastic wave simulations is limited by the Courant condition and material heterogeneity. Its computational efficiency is significantly hampered when the mesh size is much smaller than a wavelength (for geometric modeling accuracy) and/or with a high impedance contrast. An unconditionally stable alternating direction implicit (ADI) method is proposed to overcome the conditional stability. It is based on additively splitting the Crank Nicholson (CN) operator into two sub-operators and subsequently solving the CN scheme in two sub-steps. In each substep, a tri-diagonal matrix is formed based on one of the sub-operators and the associated field variables are solved implicitly with O(N) computational complexity. The rest of the field variables are solved explicitly. Due to its semi-implicit nature, it can be proved that the ADI method is unconditionally stable regardless of the time step size. The numerical dispersion properties of the ADI method are also analyzed theoretically. Several numerical examples of acoustic wave scattering from elastic targets are further provided to demonstrate its accuracy and efficiency.

3:15–3:30 Break

3:30

4pEA10. Movable optical lens array using acoustic radiation force. Daisuke Koyama, Yuta Kashihara (Faculty of Sci. and Eng., Doshisha Univ., 1-3 Tataramiyakodani, Kyotanabe, Kyoto 610-0321, Japan, dkoyama@mail.doshisha.ac.jp), Megumi Hatanaka (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan), Kentaro Nakamura (Precision and Intelligence Lab., Tokyo Inst. of Technol., Yokohama, Japan), and Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Japan)

A movable optical lens array that utilizes acoustic radiation force was investigated. The lens array consists of a rectangular glass plate, two piezoelectric bimorph transducers, and a transparent viscoelastic gel film. By electrically exciting the transducers, the flexural vibration mode was generated along the glass plate. The acoustic radiation force acts to the surface of the gel so that the surface profile can be deformed and the lens array can be fabricated. A cylindrical lens array with the lens pitch of 4.6 mm was fabricated at 113 kHz. The lens positions correspond with the loop positions of the flexural standing wave, and the lens pitch corresponds with half the

wavelength of the flexural vibration of the glass plate. The focal points of the lens could be changed by the input voltage, and the lens can act as a variable-focus lens. The lens positions could be moved in the length direction by two-phase drive. The moving distance of the lens was 4.6 mm when the driving phase difference between the two transducers changed from 0 to 360°. The translation of the lens array was applied as an optical scanner.

3:45

4pEA11. Thermoacoustic energy harvesting. Andrew W. Avent and Christopher R. Bowen (Mech. Eng., Univ. of Bath, Claverton Down, Bath, Sommerset BA2 7AY, United Kingdom, a.avent@bath.ac.uk)

Thermoacoustics have a key role to play in energy harvesting systems, exploiting a temperature gradient to produce powerful acoustic pressure waves. As the name suggests, thermoacoustics is a blend of two distinct disciplines: thermodynamics and acoustics. The field encompasses the complex thermo-fluid processes associated with the compression and rarefaction of a working gas as an acoustic wave propagates through closely stacked plates in the regenerator of a thermoacoustic device; and the acoustic network that controls the phasing and properties of that wave. Key performance parameters and appropriate figures of merit for thermoacoustic devices are presented with particular emphasis upon the critical temperature gradient required to initiate the acoustic wave and the thermal properties of the key component, namely, the "stack" or "regenerator". Mechanisms for coupling a thermoacoustic prime mover with electromagnetic harvesters and piezoelectric transducer materials are also presented, which offer the potential to enhance the energy density attained beyond that possible with linear alternators. Numerical modeling strategies are presented, which enable parametric sweeps of the geometric and thermal properties, which influence the efficiency, and performance of the key components of such devices. Potential coupling and non-linear effects are examined.

4:00

4pEA12. A unified equivalent circuit of electromechanical transducers. Li-Feng Ge (School of Elec. Eng. and Automation, Anhui Univ., 111 Jiulong Rd., Hefei 230601, China, lf_ge@hotmail.com)

It has been interesting to find a unified equivalent circuit of electromechanical transducers for a long time. Hunt provided a method using a space operator to represent all transducer types with a single form of equivalent circuit [Hunt, Electroacoustics, 1954]. But, as indicated by Beranek, the space operator k does not commute with the time operator j, so that one must define $jk=-kj$ [Beranek, JASA **77**(4), 1309–1313, 1985]. During the last decades, the research on micromachined electrostatic or capacitive ultrasonic transducers stimulates further the interest in the subject [Ge, Sc. China, A **29**(11), 1308–1315, 1999]. In this work, the space operator is not used. A transformation factor is defined as a ratio of blocked electrical-to-mechanical transfer impedance to blocked electrical impedance, thus, the factor is a real number for reciprocal transducers and imaginary for antireciprocal ones, which indicates the essential difference of the two types of transducers. Then, the corresponding driving-point electrical impedances are derived, and the expressions for the two are identical, demonstrating their common properties also. Thus, a unified impedance-type equivalent circuit for electromechanical transducers can be further obtained. [Work supported by NSFC (Grant Nos. 60374044 and 60774053).]

4:15

4pEA13. Fabrication of micro-lens array using surface acoustic wave. Satoki Taniguchi (Faculty of Life and Medical Sci., Doshisha Univ., Tataramiyakodani 1-3, Kyotanabe, Kyoto 610-0321, Japan, dmo1038@mail4.doshisha.ac.jp), Shinji Takayanagi, Daisuke Koyama (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Kyoto, Japan), Kentaro Nakamura (Precision and Intelligence Lab., Tokyo Inst. of Technol., Yokohama, Japan), and Mami Matsukawa (Faculty of Sci. and Eng., Doshisha Univ., Kyotanabe, Japan)

A technique to form an optical micro-lens array using surface acoustic wave (SAW) was investigated. The lens has no mechanical moving parts, such as gearing systems, and is composed of a simple structure. A viscoelastic transparent silicone gel film with the thickness of 20 μm was formed on a 128°-rotated Y-cut X-propagation $LiNbO_3$ (LN) substrate between two interdigitated transducers (IDTs). The IDT electrodes consisted of 30 finger pairs with the aperture length of 9.1 mm and the periodic pitch of 200 μm were fabricated on the LN substrate. The IDTs was excited with a continuous sine wave with the amplitude of 0 to 15 V_{pp} at the resonance frequency (20 MHz) to generate the SAW. The leaky SAW propagated into the gel, and the acoustic radiation force acted to the surface of the gel film so that the surface profile of the gel could change and a micro-lens array could be fabricated on the gel. The lens height could be controlled by varying the voltage applied to the IDTs.

4:30

4pEA14. *PT*-symmetric acoustics. Xuefeng Zhu, Hamidreza Ramezani, Chengzhi Shi, Jie Zhu, and Xiang Zhang (Dept. of Mech. Eng., Univ. of California, Berkeley, 3112 Etcheverry Hall, Berkeley, CA 94720, chengzhi.shi@berkeley.edu)

The concept of acoustic parity-time (*PT*) symmetry is introduced and used for the study of extraordinary scattering behavior in acoustic *PT*-symmetric media consist of loss and gain units. The analytical study of acoustic *PT*-symmetric media shows that these media can be designed to achieve unidirectional transparency at specific frequencies named exceptional points (EPs). This unidirectional transparency occurs at the EPs is due to the asymmetrical arrangement of the periodic loss and gain units that results in different Bragg scatterings on the two sides of the *PT*-symmetric media. A close look at the phases of the reflections on both sides reveals a sudden jump of the reflection phase on one side at the EPs. This step-function like behavior causes an infinite delay time of the reflected wave on that side, and hence the media become reflectionless in that direction. Combining the concept of acoustic *PT*-symmetry with transformation acoustics, we design a two-dimensional acoustic cloak that is invisible in a prescribed direction. This kind of directional cloak is important especially for military use since a target object is hidden from the enemy in front can still be identified by friendly at the back. Other useful applications such as directional acoustic imaging, noise cancellation, architectural acoustics, acoustic amplification, etc., can also be developed.

4:45

4pEA15. Smoothed particle hydrodynamics approach for modeling sound of a rigid body falling on water. YongOu Zhang, Tao Zhang, TianYun Li, and Peng Wang (Dept. of Naval Architecture and Ocean Eng., Huazhong Univ. of Sci. and Technol., East 2nd Bldg., Wuhan, Hubei 430074, China, zhangyo1989@gmail.com)

Computational acoustic methods based on Eulerian description are widely used in industrial applications. However, some special acoustic problems, such as transient acoustic problems with moving or deformable boundaries, object separation, or for multiphase systems, are still cannot ideally solved with these Eulerian methods. The present work aims at using a Lagrangian meshfree method, the smoothed particle hydrodynamics (SPH), to simulate a time-domain acoustic problems with moving boundaries which is the sound of a rigid body falling on water. First, Lagrangian acoustic wave equations considering the sound source based on the hydrodynamic/acoustic splitting method are given and represented in the SPH form. Then, two-dimensional simulation of a rigid object falling on the free surface of water is computed by the SPH method. Noise sources are obtained from the flow field information of each fluid particle. Finally, acoustic experiments with measuring the sound of the rigid body falling on water are used to test and validate the SPH results. The accuracy and efficiency of the SPH acoustic computation are evaluated, and a comparison of cases with different impact velocities is also discussed.

5:00

4pEA16. Identification of fractures in carbonates using sonic imaging logs Example from the central of East European plain. Xiaohua Che, Wenxiao Qiao, Peng Liu, Xiaodong Ju, and Junqiang Lu (State Key Lab. of Petroleum Resources and Prospecting, College of Geophys. and Information Eng., China Univ. of Petroleum-Beijing, 18 Fuxue Rd., Changping, Bejing 102249, China, imchexh@gmail.com)

Compared with sandstone, the structure of carbonate rocks is much more complex for its heterogeneity and anisotropy. The Republic of

Tatarstan of Russia lies in the central of East European Plain, where oil-gas resource is very rich, and formations are mainly carbonates. In order to detect the underground geology of the region, array acoustic logging tool, micro-electrical scanning tool, and conventional logging tools were applied. We processed and analyzed the array acoustic logging data. The results demonstrated that monopole shear wave attenuation, Stoneley wave attenuation and monopole variable-density waveforms were all sensitive to the boundary where wave impedance contrast was high. The boundary may be a fracture, or a bed interface between shale and limestone in the region. In the micro-electrical scanning images, both the low-angle fracture and the bed interface showed one sine curve. The sine curve of the low-angle fracture was relative rough and the curve width changed with different azimuth, compared with that of the bed interface. The direction of fracture and stress can be obtained by processing cross-dipole waveforms of array acoustic logging data. Micro-electrical scanning tool provided three caliper curves of different azimuth that can determine borehole expanding direction, further getting the stress direction.

THURSDAY AFTERNOON, 21 MAY 2015 BRIGADE, 1:30 P.M. TO 4:30 P.M.

Session 4pMU

Musical Acoustics: General Topics in Musical Acoustics II

Randy Worland, Cochair
Physics, University of Puget Sound, 1500 N. Warner, Tacoma, WA 98416

Jack Dostal, Cochair
Physics, Wake Forest University, P.O. Box 7507, Winston-Salem, NC 27109

Whitney L. Coyle, Cochair
The Pennsylvania State University, 201 Applied Science Building, University Park, PA 16802

Chair's Introduction—1:30

Contributed Papers

1:35

4pMU1. Measurement and analysis of musical vibrato parameters. Mingfeng Zhang, Mark Bocko (Dept. ECE, Univ. of Rochester, Rochester, NY 14627, mzhang43@hse.rochester.edu), and James W. Beauchamp (ECE, Univ. of Illinois at Urbana-Champaign, Urbana, IL)

Vibrato is a fundamental feature of many musical sounds, with vocal, stringed instrument, and many wind instrumental performances employing vibrato as a principal device for conveying musical expression. In this paper, we describe a signal processing toolbox for extracting and analyzing vibrato-related parameters from audio recordings. In our method, sonic partials of a musical sound are separated first, followed by tracking of the instantaneous amplitude and frequency of each partial. A number of parameters that characterize vibrato can then be extracted from the tracked partials; these include the common frequency of the amplitude modulation (AM) and the frequency modulation (FM), the modulation depths, the relative phase between the AM and FM components of each partial, and the harmonic contents of the modulation components. This then enables a detailed comparison of the vibrato of various musical sounds, for example the relative amounts of AM and FM, the rate of vibrato, the correlation and phase between AM and FM vibrato components, and many other features. This framework provides a useful tool for applications such as music performance pedagogy and musicological studies.

1:50

4pMU2. Real-time visualization of musical vibrato for music pedagogy. Minhao Zhang, Mingfeng Zhang, Sarah Smith (ECE, Univ. of Rochester, Rochester, NY 14627, mzhang46@ur.rochester.edu), and Mark Bocko (ECE, Univ. of Rochester, Rochester, NY)

Vibrato is a fundamental expressive attribute in music, especially in singing, in stringed instrument performance, and in the performance techniques of many wind instruments. Performers typically invest a great deal of time and practice to gain adequate control of vibrato in performance To assist and accelerate this learning process we are developing computer based vibrato visualization tools. In this paper, we present a low-latency, near real-time system that enables performers to visualize their vibrato. The system employs a video-game-like visualization scheme to display the instantaneous AM/FM trajectories of a musical sound, either by itself or in comparison to pre-recorded target sounds, to enable the user to attempt to match the vibrato trajectory of an existing performance. In addition to demonstrating the system, feedback from music students and studies of the effectiveness of the system will be presented.

2:05

4pMU3. Listener preferences for vibrato rate and extent in synthesized vocal samples. John P. Nix (Music, Univ. of Texas San Antonio, One UTSA Circle, San Antonio, TX 78249, john.nix@utsa.edu)

An experiment was devised to test the preferences of singers, vocal teachers and choral directors for vibrato rate and extent on the /a/ and /i/ vowels and with two different "gender" singers by using the voice synthesis program Madde, which is capable of varying vibrato rate, extent,

fundamental frequency, and formant frequencies. The study also sought to determine if the vowel and "gender" of the singer (as determined by the formant frequencies used for the simulation) have an impact on listener preferences. Subjects answered a series of questions about their musical training and occupation, then listened to four samples each, two of the "male" voice performing the vowels at 220 Hz, and two of the "female" voice performing the same vowels at 440 Hz. The order in which the samples were presented and the initial vibrato rate and extent conditions of the simulations were varied to try to negate any order effects on the data. The investigator adjusted the vibrato rate and vibrato extent until the listener felt his or her preference was being best represented. Listeners displayed a wide variety of preferences, with those most involved in musical theater and early music performance preferring a narrower vibrato extent.

2:20

4pMU4. Perception of non-vibrato sung tones. Randi Wooding and John P. Nix (Music, Univ. of Texas San Antonio, One UTSA Circle, San Antonio, TX 78249, john.nix@utsa.edu)

Singers are often asked to sing with a non-vibrato production. However, the term non-vibrato is problematic, as all human singing involves fundamental frequency variation. Whether a singer achieves a quality of tone that is perceived as non-vibrato may depend upon the experience of the listener. The specific aim of this study was to determine at what point a tone is perceived as non-vibrato by a population (N = 131) of singers, voice teachers, choir directors, and speech pathologists. Utilizing voice samples that exhibit a variety of vibrato rates and extents, the investigators sought to determine: (1) if there is a threshold for the perception of non-vibrato tone with regards to vibrato extent; (2) if given similar vibrato extent, does vibrato rate effect perceptual judgments of non-vibrato tone; (3) if there are differences in the perceptual threshold of non-vibrato tone across different professions of listeners. Participants responded to an online survey featuring randomized samples of sopranos singing [ɑ] with a variety of vibrato rates and extents. Some samples were repeated to test response reliability. Results indicate a perceptual threshold exists with regards to vibrato extent, and that vibrato rate can override the effect of a narrow or wide extent to some degree.

2:35

4pMU5. Estimation of a violin source/filter model. James W. Beauchamp (Dept. of Elec. and Comput. Eng. and School of Music, Univ. of Illinois at Urbana-Champaign, 1002 Eliot Dr., Urbana, IL 61801-6824, jwbeauch@illinois.edu) and Ashwin K. Vijayakumar (Dept. of Electronics and Commun. Eng., National Inst. of Technol., Karnataka, Surathkal, Karnataka, India)

The source/filter model is often mentioned in the acoustics and signal processing literature (e.g., Gold and Morgan, *Speech and Audio Signal Processing*, Wiley) but has seldom been implemented for musical instrument sounds. An exception was by Mathews and Kohut, ("Electronic simulation of violin resonances," J. Acoust. Soc. Am. **53**(6) (1973)). In this study, we attempt to separate source harmonic amplitudes from a violin filter characteristic using harmonic tracking analysis of glide tones performed in an anechoic chamber. Assuming that the violin bridge/body constitutes a linear system, if string force harmonics are constant over pitch, they should sweep out frequency responses that only differ by constant amplitude ratios over their overlap regions. Departures from constant ratio behavior can be interpreted in terms of source spectra that change with fundamental frequency. Interpretation and detailed results of this analysis are given.

2:50

4pMU6. The influence of sung vowels on pitch perception. Johanna Devaney and Derek Richardson (School of Music, The Ohio State Univ., 1899 N. College Rd., Columbus, OH 43210, devaney.12@osu.edu)

Phonologists have demonstrated that in speech there is an intrinsic pitch of vowels, i.e., that different pitch heights are perceived for different spoken vowels with the same mean fundamental frequency. Only limited work has been done on the impact of sung vowels on perceived pitch and previous studies, using forced choice paradigms, have found conflicting results comparing Front/Close to Back/Open vowels: Ternstrom, Sundberg, and Colldén (1988), using manipulated sung tones, found that Front/Close vowels were perceived as higher than other vowels while Fowler and Brown (1997), using synthesized tones, found that Front/Close was perceived lower than Back/Open. The current study uses a method-of-adjustment paradigm to examine pitch perception of all four extremal vowels: Front/Close (/i/), Front/Open (/æ/), Back/Close (/u/), and Back/Open (/ɑ/), using completely controlled, but realistic synthesis. It also assesses whether different models of pitch perception are necessary for the various vowel types. In the experiment, the subjects are asked to match a reference tone that has no formants against a stimulus tone with first and second formants corresponding to a particular vowel. Both the reference and stimulus tones are identical in their synthesis except for the formants and are 300 ms in length.

3:05–3:15 Break

3:15

4pMU7. Effect of acoustic feedback on the singing voice. Pasquale Bottalico (Dept. of Communicative Sci. and Disord., Michigan State Univ., Via Sant'Anselmo 32, Torino 10125, Italy, pasqualebottalico@yahoo.it), Eric J. Hunter, and Simone Graetzer (Dept. of Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Voice control is of great importance in a singer's training, and in particular, control of pitch (fundamental frequency) and volume (sound pressure level). We hypothesize that a singer's vocal comfort and control will be increased as the acoustic feedback of the room is increased. In this study, 20 singers (10 amateur and 10 professional singers) performed vocal tasks in different acoustic conditions. Vocal tasks comprised scales and arpeggios with different dynamics at different speeds, and an extract from the American National Anthem, which was accompanied by a musical track emitted at different power levels. After each condition, the subject responded to questions regarding perception of vocal comfort, control, and fatigue, and their own voice feedback. Room acoustics were manipulated: in some conditions, two reflective panels were placed at different distances from the subject. The results indicate that when panels were present, singers tended to perceive the room as being more pleasant to sing in. Intonation (pitch control) and the Lombard Effect (a tendency to increase in voice level as background noise level increases) for singers are related to variation in the room acoustics.

3:30

4pMU8. Differentiated electroglottograph and audio signal measurements of vocal fold closed quotient during a register change: Single note data. Richard J. Morris (Commun. Sci. and Disord., Florida State Univ., 201 West Bloxham Rd., 612 Warren Bldg., Tallahassee, FL 32306-1200, richard.morris@cci.fsu.edu), David Okerlund (College of Music, Florida State Univ., Tallahassee, FL), and Shonda Bernadin (College of Eng., FAMU/Florida State Univ., Tallahassee, FL)

Recently, the use of the differentiated electroglottograph (dEGG) signal combined with a differentiated audio (dAudio) signal was reported as a means for more reliable measurement of the closing quotient (CQ) from the EGG signal in combination with a time synchronized audio signal. The purpose of this study is to combine CQ data with resonance data from the vocal tract during a register shift. Files recorded from group of 15 trained females provided the data. The singers were directed to shift between chest and mixed registers while sustaining a note that could be sung comfortably in both registers. An EGG and an audio signal were recorded. These signals were time aligned, and the EGG, dEGG, and dAudio were displayed. These three signals were used to determine the CQ and peak resonant amplitude before and after the register shift. Preliminary results indicate that the singers maintained a similar CQ measurement across the register shift. In addition, the preliminary data indicate that the singers changed the dominant harmonic in the sung note when they changed registers. That is, they maintained the same fundamental frequency but altered the resonance of the tone in the vocal tract.

3:45

4pMU9. Comparing time varying directivity of musical instruments across different musical motifs. Madeline A. Davidson, Kristin Hanna, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, madeline.davidson@huskers.unl.edu)

The directionality of a musical instrument can vary in time quite significantly as the instrument is used to play a musical piece, but this behavior has not been well-quantified or compared across instruments and musical motifs in previous research. Prior work conducted at the University of Nebraska has proposed quantifying this behavior by seeing how directional characteristics vary in time using overlapping one-second windows. The analyses were conducted on twenty-second long sections from 13-channel anechoic recordings of three solo instruments (flute, violin, and trombone). Maximum directivity index plots across time and space were produced to quantify and examine the time-varying directional characteristics. In this presentation, these previous results are compared against more recent results obtained from five-channel anechoic recordings of all instrument parts of both a Mozart and a Brahms symphony. The effects of reducing the time window of analysis from one second down to half-second and quarter-second time windows are also examined. Special attention is paid to how the proposed quantifiers for time-varying directivity are impacted by aspects such as tempo and dynamics in these musical compositions.

4:00

4pMU10. Effects of pitch, intensity, and timbre on frequency masking. Song Hui Chon and David Huron (Ohio State Univ. School of Music, 1866 College Rd., Columbus, OH 43210, chon.21@osu.edu)

It has been widely known that the four main factors of music—pitch, timbre, duration, and loudness—are not independent. One comes to wonder how these four factors will affect frequency masking, which has traditionally been explained with the cochlear responses to pure tones or narrowband noises. As a first step toward understanding these interactions, the mutual masking effects of two concurrent non-unison tones were investigated. A MATLAB simulation was implemented based on the psychoacoustic model in Bosi & Goldberg (2003). As the degree of masking would obviously depend on factors such as pitch, timbre, and intensity, we considered 81 pitches (B0 to G7) in three timbre categories (pure tone, uniform, and average timbre). For each timbre category, 15 levels of amplitudes were considered to examine the effect of intensity. A preliminary analysis shows that the chance of the higher pitched tone masking the lower pitched tone might change depending on the pitch, timbre, and loudness. The findings will be discussed with the implications on the interpretation of high voice superiority on the preference of melodies on the highest part in music.

4:15

4pMU11. Demonstration of nonlinear mode coupling in woodwind-like air columns: Recollections from the laboratory of A. H. Benade. Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ., N-249 Millennium Sci. Complex, University Park, PA 16803, sct12@psu.edu) and Peter L. Hoekje (Phys. and Astronomy, Baldwin Wallace Univ., Berea, OH)

Self sustained oscillations in woodwinds are generated by the player's blowing pressure, through the interaction of the standing wave in the air column, the dynamic motion of the reed and the air flow through the reed. In properly designed instruments, the oscillation begins at a threshold blowing pressure near the frequency of an air column resonance. With increased blowing pressure, the amplitude grows and the flow nonlinearity generates harmonics of the playing frequency. The nonlinearity couples the steady state components of the tone. The coupling dictates behaviors that cannot be explained by a linear model. When A. H. Benade came to understand this in the early 1960s, he developed a "tacet horn" whose unplayable behavior could only be explained by the nonlinear theory [Benade and Gans, Ann. N.Y. Acad. Sci. **155**, 247–263 (1968)]. Throughout his life, Benade used other pathological air columns for research and pedagogy. This paper will demonstrate two of Benade's concepts with air columns that are simple to build. One is a clarinet-like tube that can play softly near threshold, but cannot be played louder. The other is a conical tube that plays only at high amplitude.

THURSDAY AFTERNOON, 21 MAY 2015 — KINGS 5, 2:00 P.M. TO 5:00 P.M.

Session 4pNS

Noise, ASA Committee on Standards, Psychological and Physiological Acoustics, Physical Acoustics, and Structural Acoustics and Vibration: Wind Turbine Noise II

Robert D. Hellweg, Cochair
Hellweg Acoustics, Wellesley, MA 02482

Kenneth Kaliski, Cochair
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Paul D. Schomer, Cochair
Schomer and Associates Inc., 2117 Robert Drive, Champaign, IL 61821

Nancy S. Timmerman, Cochair
Nancy S. Timmerman, P.E., 25 Upton Street, Boston, MA 02118

This session builds on the session 4aNS at which research and technical background useful toward the development of the wind turbine acoustic emissions agenda will be presented. Session 4pNS will consist of panelists along with audience participation with the purpose of developing the start of a draft research agenda. Invitees, to the best of our abilities, will include all interested parties, including relevant agencies from the United States and Canada, medical practitioners, other research professionals, and acoustical consultants with wind turbine acoustic emissions experience.

Session 4pPA

Physical Acoustics: Infrasound II

Roger M. Waxler, Chair
NCPA, University of Mississippi, 1 Coliseum Dr., University, MS 38677

Contributed Papers

2:00

4pPA1. Propagation model based explosive yield determination from stratospheric infrasound arrivals: Humming Roadrunner data analysis. Roger M. Waxler (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu), David Green (AWE Blacknest, Reading, United Kingdom), and Jean-Marie Lalande (NCPA, Univ. of MS, University, MS)

Attempts to estimate explosive yields from stratospheric infrasonic returns have often relied on empirical relationships between explosive charge weight, source-to-receiver range, and a simple parameterization of the atmospheric state. We propose to approach the yield estimation problem by combining knowledge of acoustic generation by explosions and numerical acoustic propagation modeling using state-of-the-art atmospheric specifications (G2S). The goal is to provide a more accurate and robust yield estimation tool that can incorporate a variety of meteorological scenarios. We introduce an approach based on modeling the transmission loss between source and receiver using an incoherent modal sum. The total observed signal power is related to the near-field acoustic power generated by the source event. The technique is applied to stratospheric returns recorded from the Humming Roadrunner Ground Truth experiments performed in the August 2012 in the American West.

2:15

4pPA2. Theoretical model and experimental validation of a gas-combustion infrasound source. Chad M. Smith and Thomas B. Gabrielson (Graduate Program in Acoust., Appl. Res. Lab., The Penn State Univ., State College, PA 16804, cms561@psu.edu)

The design of controlled infrasound sources is challenging due to the large volumes of air that must be moved with a source of practical size. One possible approach is to produce a periodic volume expansion/contraction by periodic heating of the air mass using a combustible-gas burner. For equal storage pressure, the available energy density from combustion of a gas is substantially higher than the available energy density from expansion of a compressed gas. While no source that is much smaller than a wavelength will be anything but inefficient, this large energy density may partially offset this basic acoustical inefficiency making gas-combustion infrasound source technology practical for use in the field. The goal of this work is to generate usable levels of sound in the 0.1 to 20 Hz frequency band for infrasound-measurement-system calibrations and propagation-path characterization. This paper presents a theoretical model and experimental demonstration of this infrasound generation method. Both theory and demonstration suggest that usable infrasound levels can be generated with a relatively compact source.

2:30

4pPA3. The validation of SSWs forecasts in atmospheric global circulation models. Jelle D. Assink (DAM/DIF/DASE, CEA, Bruyeres-le-Chatel, Arpajon F-91297, France, jelle.assink@gmail.com), Pieter Smets, Läslo Evers (KNMI, De Bilt, Netherlands), and Alexis Le Pichon (DAM/DIF/DASE, CEA, Arpajon, France)

While the influence of the troposphere on the stratosphere is well known, recent observational and modeling studies have demonstrated that the stratosphere has an impact on the troposphere as well. The dynamical coupling between stratosphere and troposphere is particularly strong during sudden stratospheric warming (SSW) events. The correct forecasting of the onset and duration of SSW events is therefore important and is a current challenge for weather forecasting centers. As there is a lack of observations in the upper stratosphere with good temporal and spatial coverage, additional techniques may be helpful to constrain SSWs. This is illustrated using volcanic infrasound measurements. The observations are compared with nowcast and forecast models up to 10 days. While a general agreement is found during the summer period, larger discrepancies are found during the equinox and major SSW of January 2013.

2:45

4pPA4. Status of digital infrasound sensors developed by the NCPA. Carrick L. Talmadge (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38655, clt@olemiss.edu)

In collaboration with Hyperion Technology Group, Inc., the National Center for Physical Acoustics (NCPA) has developed a digital infrasound sensor that can be configured for broadband outdoor measurements (flat within 3-dB from 0.03–150 Hz) and a nominal maximum transducible pressure of 200-Pa peak-to-peak, ultra-broadband measurements for calibration systems (flat within 3-dB from 0.0003-Hz to 150 Hz) and very high level sounds (up to 110 kPa peak-to-peak). This sensor has a GPS-locked digitizer that store over four months of continuously sampled data digitized at 1000 samples per second. The measurement performance of this system, including noise floor, reproducibility of measurements between sensors, linearity, mechanical robustness, etc., will be summarized.

3:00

4pPA5. The interaction between infrasonic waves and gravity waves perturbations: Application to explosions at the Utah test and training range. Jean-Marie Lalande and Roger Waxler (National Ctr. for Physical Acoust., Univ. of MS, 1 Coliseum Dr., University, MS 38677, jlalande@olemiss.edu)

Infrasonic waves propagate at long range through atmospheric ducts resulting from the stratification of atmospheric properties. These ducts are characterized by their spatio-temporal variability. Hence, infrasonic waves integrate information upon the atmosphere along their propagation paths. In order to study infrasonic wave propagation, we resort to atmospheric specification combining numerical weather prediction and climatological models. However, due to the lack of observations these models fail to describe small scale variability such as perturbations associated to the presence of internal

gravity waves. These waves play an important role in the atmospheric dynamic by transferring momentum to the mean flow at critical levels and at wave-breaking altitudes. In this study, we intend to describe the interaction of infrasonic waves with internal gravity waves in order to understand the long-tail behavior observed in infrasound broadband signals. We developed a model for the propagation of internal waves used to generate realistic perturbations of the background atmospheric states. By using a linear full-wave model of infrasound propagation, our goal is to ultimately relate infrasound characteristics to internal waves properties. We apply those numerical models to infrasound observations recorded during the 2010 deployment of the Utah Test and Training Range (UTTR) experiment.

3:15

4pPA6. On atmospheric updating, attenuation, and thermospheric phases in infrasonic wave-trains. Roger M. Waxler (NCPA, Univ. of MS, 1 Coliseum Dr., University, MS 38677, rwax@olemiss.edu), Jelle Assink (CEA, Paris, France), and Joel Lonzaga (NCPA, Univ. of MS, University, MS)

Because of the dramatic increase in temperature in the thermosphere, thermospheric phases are ubiquitous, despite the fact that they are less often detected than are stratospheric phases. In the thermosphere, the density of the atmosphere is very low so that, as a propagation medium, the thermosphere is very non-linear and very attenuating. Atmospheric specifications of the lower thermosphere are not well constrained by data. This can lead to significant errors in modeling propagation along thermospheric paths. Methods for generating corrections to specifications of the thermosphere from infrasonic data are described. Further, using a non-linear ray theory model we find that non-linear effects in the thermosphere, in particular, period lengthening, mitigate against attenuation. Finally, we show that it is possible to use thermospheric returns to simultaneously update the atmospheric specifications while determining source yield.

3:30

4pPA7. Infrasonic propagation modeling of orbital launch vehicles. John M. Noble, W. C. K. Alberts, and Stephen M. Tenney (Army Res. Lab., 2800 Powder Mill Rd., Adelphi, MD 20783, john.m.noble.civ@mail.mil)

Over the last two years, NASA's Wallops Flight Facility has been launching medium lift rockets for experimental and space station resupply missions. These launches have been a great opportunity to use the rocket-generated infrasound as a repeatable source to study the long range propagation over different seasons. For some of the launches during this period, two, 20-m arrays were deployed along different directions from the launch site. To increase the number of available sensors for comparison to three-dimensional propagation model results, data from the US Array was incorporated into the study. The US Array data significantly increased the range and azimuths used to compare the propagation modeling and measurement results. Calculated and measured transmission losses will be discussed.

THURSDAY AFTERNOON, 21 MAY 2015 KINGS 4, 1:30 P.M. TO 5:15 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Physiology, Behavior, and Modeling: From Middle-Ear to Mid-Brain

Sunil Puria, Chair
Stanford University, 496 Lomita Mall, Stanford, CA 94305

Contributed Papers

1:30

4pPP1. Effect of cueing on stability of behavioral measurements of basilar membrane responses with a precursor. Thibaud Necciari and Bernhard Laback (Acoust. Res. Inst., Austrian Acad. of Sci., Wohllebengasse 12-14, 1st. Fl., Vienna A-1040, Austria, thibaud.necciari@oeaw.ac.at)

Stimulation of the olivocochlear efferent system can reduce the gain of the basilar membrane (BM) response to a sound. This gain reduction may be accompanied by a reduction in BM compression, although behavioral data are inconclusive. In this study, the effect of efferent stimulation on compression is assessed using fixed-duration masking curves (FDMCs) with a precursor. In a pilot experiment, presenting a broadband noise precursor before the masker resulted in large within-listener variability for the shortest target durations, although a three-interval forced-choice task was used. To reduce possible confusion effects and increase the dynamic range of BM response measurement, a contra-lateral broadband cue signal, gated simultaneously with the precursor and masker, was added. FDMCs with a 150-ms precursor were measured with and without cue for target durations of 3, 9, and 15 ms. The total masker-target duration was fixed at 25 ms. Listeners were trained at least 2 h, then each condition was measured three times. In contrast to the literature, the presence of the cue neither reduced within-listener variability nor increased sensitivity. Alternative methods are thus required to improve the stability and dynamic range of behavioral measurements of BM responses with a precursor. [Work supported by the FWF project I-1362-N30.]

1:45

4pPP2. Distortion-product otoacoustic emission fine structure as an indicator of combination-tone-mediated masking release. James D. Lewis, Emily C. Clark, Judy G. Kopun, Walt Jesteadt, Stephen T. Neely, and Michael P. Gorga (Boys Town National Res. Hospital, 555 N 30th St., Omaha, NE 68131, james.lewis@boystown.org)

During simultaneous masking, when the probe (fp) is higher in frequency than the masker (fm), the cubic difference tone (CDT; $f_{cdt} = 2f_m - f_p$) can provide masking release. Given their shared origin, the 2f1-f2 distortion-product otoacoustic emission (DPOAE) may be a physiological correlate of CDT-mediated masking release [McFadden *et al.* (2012). J. Acoust. Soc. Am. **132**, 968–983]. To test this hypothesis, masking release and DPOAEs were measured in normal-hearing listeners for 2 and 4 kHz probes. Probe threshold was measured in the presence of a masker ($f_p/f_m = 1.2$) and masker-suppressor pair. The suppressor was designed to mask detection of the CDT. Masking release was calculated as the change in masked-probe threshold resulting from presentation of the suppressor. DPOAEs were measured for an f2 ($f_2/f_1 = 1.2$) between 1.8 and 2 kHz (6-Hz intervals) and between 3.6 and 4.4 kHz (12-Hz intervals). Masking release was associated with DPOAE fine structure. Listeners without masking release lacked DPOAE fine structure. Fine structure was due to wave interference between distortion- and reflection-source components; the reflection-source component is thought to originate from the place of CDT detection. Findings suggest a similar underlying cochlear mechanism for the perceptual and acoustical effects. [Work supported by NIH-NIDCD grants 1F32DC014175-01A1, 5R01DC002251-18, 2R01DC008318-06A1, and 5P30DC004662-13.]

2:00

4pPP3. Can auditory brainstem and midbrain processing of interaural level difference cues really explain perceptual performance? Stephanie T. Cheung and Ian C. Bruce (Elec. & Comput. Eng., McMaster Univ., Rm. ITB-A213, 1280 Main St. W, Hamilton, Ontario L8S 4K1, Canada, ibruce@ieee.org)

Neurons in the lateral superior olive (LSO) are believed to be involved in processing of interaural level difference (ILD) cues for sound localization. However, the ILD-tuning of LSO neurons varies with the absolute sound pressure levels (SPLs) at the two ears, in contrast to the relatively robust perceptual processing of ILD cues at different absolute SPLs. Tsai *et al*. (J. Neurophysiol. 2010) proposed that if some inferior colliculus (IC) neurons compute the difference between the contralateral and ipsilateral LSO outputs, the dependencies on absolute SPLs could be canceled out. However, they only considered pure-tone stimulation of single neurons at their characteristic frequencies and at relatively low SPLs. In this study, we evaluated the hypothesis of Tsai and colleagues using computational models of populations of auditory nerve, LSO and IC neurons consistent with Tsai and colleagues' proposal. Predictions from two different neural decoder algorithms applied to LSO and IC model outputs indicate that discharge-rate saturation and spread of excitation in the auditory nerve lead to even greater SPL-dependence of ILD coding across a population of LSO neurons than is apparent in single neurons, and the IC differencing operation proposed by Tsai and colleagues does relatively little to ameliorate this problem.

2:15

4pPP4. Effect of low-intensity highpass noise on stimulus frequency otoacoustic emission group delays for low frequency evoking tones. Jordan A. Beim and Magdalena Wojtczak (Psych., Univ. of Minnesota, N218 Elliott Hall, 75 East River Rd., Minneapolis, MN 55455, beimx004@umn.edu)

Stimulus-frequency otoacoustic emissions (SFOAEs) have been used in previous research to estimate cochlear tuning in humans. These estimates of tuning rely on the theory that SFOAEs arise from coherent reflections from the place on the basilar membrane (BM) with the characteristic frequency (CF) of the tone evoking the emission. Theories underlying SFOAE generation are still the subject of much debate, and several recent studies have shown evidence supporting an alternative theory postulating that generators of SFOAE are distributed basally to the CF place on the BM. Basally distributed emission generators could explain why SFOAE-derived group delays measured in the chinchilla cochlea for low-frequency tones were shorter than predicted by coherent-reflection hypothesis. The aim of the current study is to look for potential effects of basal emission generators in SFOAE-derived measurements of group delay at low frequencies in humans. SFOAE group delays were measured at 0.5, 0.75 kHz with and without a low intensity highpass noise, used to perturb basally distributed emission generation sites. Results show that the presence of the highpass noise leads to increased group delays estimated from SFOAEs, consistent with the hypothesis that basal generators of emission influence estimates of cochlear group delay at low frequencies.

2:30

4pPP5. The influence of short term perceptual learning of pitch discrimination and modulation discrimination on subcortical envelope-following and cortical steady-state EEG responses. Bonnie Lau, Dorea R. Ruggles, Sucharit Katyal, Stephen A. Engel, and Andrew J. Oxenham (Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455, bwlau@umn.edu)

Many perceptual skills improve with training, and research suggests that long- and short-term experiences modify auditory neural structures and function. Long-term cortical and subcortical plasticity has been associated with musical training and fluency in tonal languages, and short-term training effects have been regularly observed in cortical responses. Less is known about short-term subcortical plasticity, or the simultaneous relationships between subcortical and cortical responses under training conditions. The current study examines short-term learning and neural plasticity, using behavioral measures in combination with simultaneous subcortical and cortical steady-state EEG responses. Untrained, naïve subjects were randomly assigned to groups that were trained on fundamental-frequency (F0) discrimination, amplitude-modulation rate discrimination, or visual orientation discrimination. All auditory stimuli consisted of unresolved harmonic complexes with a nominal F0 of 137 Hz that were sinusoidally amplitude modulated at 100% depth and 13-Hz rate. Simultaneous subcortical envelope-following responses (EFR) to the F0 and cortical auditory steady-state responses (ASSR) to the modulation rate were measured pre- and post-training, and changes in the magnitude of phase locking to the respective frequency components were compared to changes in behavioral measures. The methods provide a new window on subcortical and cortical plasticity and their interactions. [Work supported by NIH grant R01DC05216.]

2:45

4pPP6. The notion of "frequency clusters" in spontaneous otoacoustic emission generation. Anthony Salerno (Medical Biophys., Univ. of Toronto, Mouse Imaging Ctr., Hospital for Sick Children, Toronto, Ontario, Canada, salerno.anthony92@gmail.com) and Christopher Bergevin (Phys. & Astronomy, York Univ., Toronto, Ontario, Canada)

Normal, healthy ears can emit unprovoked low-level sounds called spontaneous otoacoustic emissions (SOAEs). These arise in a wide variety of different species (e.g., humans, lizards) despite gross morphological differences commonly thought important to the associated biomechanics. As such, theoretical models of SOAE generation differ significantly in their underlying assumptions. One model class [Vilfan and Duke (2008), Biophys. J. **95**, 4622–4630; Wit & van Dijk, 2012 JASA **132**, 918–926] uses a coupled nonlinear oscillator framework, where each element exhibits a limit cycle (with a unique characteristic frequency) and is visco-elastically coupled to its nearest neighbor. This model proposes that SOAEs arise by groups of oscillators that self-entrain into "clusters," defined simply as the frequency where the largest peak in the steady-state spectral magnitude occurs. Our study sought to computationally explore more precisely what constitutes a cluster in terms of the underlying oscillator's dynamics. We found that oscillators within a cluster exhibit relatively complicated motions and poor phase coherence. Coupled with the model's inability to reproduce realistic SOAE spectra, the biomechanical relevance of a "cluster" is called into question. Modifications to the model for lizard ears (e.g., universal coupling via the rigid basilar membrane; Bergevin & Shera (2010), JASA 2398–2409) are explored.

3:00

4pPP7. No otoacoustic evidence for a peripheral basis underlying absolute pitch. Christopher Bergevin (Phys. & Astronomy, York Univ., 4700 Keele St., Petrie 240, Toronto, Ontario M3J 1P3, Canada, cberge@yorku.ca), Larissa McKetton (Biology, York Univ., Toronto, Ontario, Canada), Victoria Stone (Commun. Sci. and Disord., Univ. of Western ON, London, Ontario, Canada), Jessica Grahn (Psych., Univ. of Western ON, London, Ontario, Canada), and David Purcell (Commun. Sci. and Disord., Univ. of Western ON, London, Ontario, Canada)

Absolute pitch (AP) is the ability to identify or produce the perceived pitch of a sound (e.g., fundamental frequency of a piano note) without an external reference. This ability is relatively rare (~1/10,000 individuals possess it) and the mechanisms underlying AP are not well understood. This study examined whether there was evidence for a peripheral (i.e., cochlear) basis for AP based upon otoacoustic emissions (OAEs). The chief motivations were that both AP and spontaneous emissions (SOAEs) appear to have genetic components and anecdotal observations of prevalence in certain populations (e.g., relatively higher incidence of both in Asians). We examined SOAEs and stimulus-frequency emissions (SFOAEs) in both control ($N = 21$) and AP ($N = 13$) normal-hearing populations. We found no substantial differences in SOAE activity between groups (e.g., no evidence for one or more strong SOAEs that could act as a cue). SFOAE phase-gradient delays, measured using several probe levels (20–50 dB SPL), also showed no significant differences. This latter observation argues against sharper frequency selectivity in AP subjects. Taken together, these data support the prevailing view that AP mechanisms arise at a processing level in the central nervous system at the brainstem or higher (e.g., optimized neural coding).

3:15–3:30 Break

3:30

4pPP8. Nonlinear dynamics of the organ of Corti, modeling both outer hair cell somatic motility and hair bundle motility. Amir Nankali (Mech. Eng., Univ. of Michigan, 1092 Louise St., Apt. 14, Ypsilanti, MI 48197, nankali@umich.edu), Aritra Sasmal, and Karl Grosh (Mech. Eng., Univ. of Michigan, Ann arbor, MI)

One of the principal questions in the cochlea biophysics is the determination of relative contributions of the two active processes, OHC somatic motility and HB motility, to the mechanics of the cochlea. Because of the difficulty of eliminating one mechanism without affecting the other, an unambiguous *in vivo* measurement differentiating their effects remains elusive. Theoretical models, therefore, have been used as one way to examine the contributions of the two active mechanisms to cochlear mechanics. In this paper, we use a physiologically based model of the mammalian organ of Corti to study the hearing active process and the relative contributions of these active forces. This local model integrates the electrical, acoustic, and mechanical elements of a cross section of the cochlea. The nonlinear dynamics of this model are studied with a special emphasis on the regions of stability and the amplification of the mechanical response arising from the active processes. [Work supported by NIH-NIDCD R01-04084 and NIH NIDCD-T32-000011.]

3:45

4pPP9. Comparing human middle-ear motion and pressure gain across idealized and progressively more realistic three-dimensional finite-element models. Kevin N. O'Connor (Mech. Eng., Stanford Univ., 496 Lomita Mall, Durand Bldg., Rm. 283, Stanford, CA 94305, kevinoc@stanford.edu), Hongxue Cai (Mech. Eng., Stanford Univ., Chicago, IL), Peter K. Gottlieb, Charles R. Steele, and Sunil Puria (Mech. Eng., Stanford Univ., Stanford, CA)

The mammalian tympanic membrane (TM) is linked to the cochlea via a flexible and circuitous three-bone ossicular chain for which much of the mass is concentrated away from the entry axis to the cochlea. As the TM area (A_{TM}) is much larger than that of the stapes footplate (A_{FP}) and the length of the malleus (L_M) is somewhat longer than the incus (L_I), the middle ear is considered to function as a pressure transformer to optimize the flow of vibrations from low-density air to high-density cochlear fluid, with an ideal pressure gain defined as $(A_{TM}/A_{FP})*(L_M/L_I)$. Even so, the reasons for this complex ossicular arrangement, as opposed to the more straightforward case of a columella directly connecting the TM to the cochlea, are not entirely clear. We explore the effects of middle-ear anatomy and material properties on ossicular motion and pressure gain by comparing the behavior of a series of 3D finite-element models ranging from idealized simple axisymmetric cases of a flat-circular or conic TM connected to a columella, all the way to an anatomically accurate three-ossicle human middle-ear model based on a micro-CT scan of a temporal bone. In doing this, we will test whether the complex 3D anatomy of the human middle ear can be shown to offer any concrete advantages in terms of pressure gain over a simpler columella design, or whether other possible reasons for this complex design are more likely. [Work supported by R01 DC05960 from the NIDCD of the NIH.]

4:00

4pPP10. Simulating the response to clicks and the generation of spontaneous otoacoustic emissions using a cochlear model. Julien Meaud (G.W.W. School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., Atlanta, GA 30332, julien.meaud@me.gatech.edu)

The response of the mammalian cochlea to acoustic clicks of low intensity is characterized by a long duration that includes several lobes of vibrations. Furthermore, some mammalian ears can spontaneously emit sounds, called spontaneous otoacoustic emissions (SOAEs), which are measurable in the ear canal. These phenomena are consequences of the active feedback by outer hair cells (OHCs) in the cochlea and give important information about cochlear biophysics. In this work, we use a computational model of the cochlea coupled to a lumped parameter middle ear model. Formulation of the model using a state-space approach allows determining the linear stability and the time-domain response of the model. In order to predict several lobes of vibrations in response to a click and the emission of SOAEs, inhomogeneities are introduced in the OHC properties. The model is shown to be in excellent with many aspects of the response of the cochlea to clicks that have been observed in experiments. Numerical results are used to demonstrate the strong link between characteristics of the frequency response and of the time-domain response. With small changes in the model parameters, a linearly stable model can become an oscillatory model that can emit SOAEs.

4:15

4pPP11. Computational modeling of distortion product otoacoustic emissions. Thomas Bowling, Kaikai Che, Charlsie Lemons, and Julien Meaud (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332, Tbowling3@gatech.edu)

Otoacoustic emissions are sounds generated by the cochlea that can be measured in the ear canal. Distortion product otoacoustic emissions (DPOAEs) are generated when the cochlea is stimulated by two primary tones and are a result of nonlinearities within the outer hair cells of the cochlea. DPOAEs are commonly used clinically to determine the health of the inner ear and scientifically to obtain noninvasive data about cochlear function. A physiologically based computational model of the mammalian ear is used to study the generation of DPOAEs. This model includes a nonlinear model of the cochlea, formulated in the time-domain and based on the finite element method and a lumped parameter model of the middle ear. Model simulations for the basilar membrane displacement, intracochlear fluid pressure, and ear canal pressure at the distortion product frequency in response to a two-tone stimulus are compared with experimental data and other modeling studies. The effects of various primary frequencies and input signal levels on the generation of DPOAEs are studied.

4:30

4pPP12. Short-term adaptation in acoustical and cochlear-implant hearing. Robert L. Smith (Inst. for Sensory Res., Syracuse Univ., 621 Skytop Rd., Syracuse, NY 33484, rlsmith@syr.edu) and Mahan Azadpour (Otolaryngol., NYU School of Medicine, New York, NY)

Sound onset and changes in sound intensity appear to play important roles in many aspects of hearing, e.g., detection, speech communication, and source identification. Perhaps for this reason the auditory system displays mechanisms that emphasize changes in sound intensity at multiple levels of neurophysiological processing. Short-term adaptation, observed in auditory-nerve responses, is perhaps the most peripheral example of onset emphasis. It can be observed in its simplest form in the auditory-nerve response to a constant-intensity tone burst where firing rate is maximum at response onset and decays to a steady-state value during the tone. Firing rate drops below spontaneous rate following tone offset, and the response to brief probe tones is reduced and gradually recovers in this post-stimulation interval. Short-term adaptation is generally assumed to occur in auditory hair cells and cochlear-implant stimulation by-passes hair cells and hence this potential source of adaptation. The results reviewed here will describe a model of peripheral short-term adaptation and how this model can be added to cochlear-implant speech processing. Preliminary results will be presented that indicate that adding short-term adaptation to cochlear-implant processing produces an improvement in speech intelligibility on the order of 5% to 10% in a variety of listening tasks.

4:45

4pPP13. Intra-subject variability in frequency-following responses and cortical event-related responses to Mandarin tones. Zilong Xie and Bharath Chandrasekaran (Dept. of Commun. Sci. & Disord., The Univ. of Texas at Austin, 2504A Whitis Ave. (A1100), Austin, TX 78712, xzilong@gmail.com)

Frequency-following responses (FFRs) reflect phase-locked responses from neuronal ensembles within the auditory brainstem and midbrain. Due to the high fidelity to stimulus characteristics, the FFRs have been extensively studied as biomarker, reflecting the integrity of subcortical encoding of complex auditory events. More recent studies suggest that the FFRs are highly malleable, and are influenced by higher-level attentive and cognitive mechanisms, as well as short-term auditory experiences. Here, we evaluate intra-subject variability of FFRs to complex speech sounds and examine relative

differences in intra-subject variability between FFRs and cortical evoked responses. We elicited FFRs to two Mandarin tones (tone 1 and tone 2) from four native Mandarin speakers over five repeated daily sessions within a week. We also recorded cortical evoked responses to these two Mandarin tones presented in an oddball paradigm from these participants during these five sessions. Several measures were used to quantify FFRs: signal-to-noise ratio, stimulus-to-response correlation, and response- to-response correlation. Our analyses show that FFRs exhibited low intra- subject variability, whereas cortical responses such as N1, P2, and mismatch-negativity demonstrated higher intra-subject variability. Our findings demonstrate high individual stability in the FFRs and FFRs can be reliable biomarker to study the integrity of subcortical encoding of speech sounds.

5:00

4pPP14. Variability in word recognition by adults with cochlear implants: The roles of perceptual attention versus auditory sensitivity. Aaron C. Moberly (Otolaryngol., The Ohio State Univ., 915 Olentangy River Rd., Ste. 4000, Columbus, OH 43212, Aaron.Moberly@osumc.edu)

Enormous outcome variability exists for postlingually deafened adults who receive cochlear implants (CIs). This variability is not purely a result of the degradation of speech representations through implants. Data from two studies are presented that examined variability in word recognition as explained by "perceptual attention" and "auditory sensitivity" to acoustic cues underlying speech perception. In each experiment, postlingually deafened adults with CIs and normal-hearing (NH) controls performed three tasks: (1) word recognition in quiet; (2) labeling for three sets of stimuli that varied based on acoustic cues in the amplitude, duration, or spectral domains; and (3) discrimination of nonspeech analogs of these acoustic cues. Word recognition was found to vary widely among CI users (20 to 96%). Attention to spectral cues, as measured by weighting coefficients computed from labeling functions, beyond simple sensitivity to spectral cues, as measured by d′ values, predicted variability in word recognition. Attention and sensitivity to nonspectral cues did not predict variability in word recognition. Efforts should be made to better represent spectral cues through implants; however, facilitating attention to these cues through auditory training should also be a clinical focus.

THURSDAY AFTERNOON, 21 MAY 2015 — KINGS 3, 2:00 P.M. TO 4:05 P.M.

Session 4pSA

Structural Acoustics and Vibration, Education in Acoustics, and Physical Acoustics: Demonstrations of Structural Acoustics and Vibration

Daniel A. Russell, Chair
Graduate Program in Acoustics, Pennsylvania State University, 201 Applied Science Bldg., University Park, PA 16802

Chair's Introduction—2:00

Invited Papers

2:05

4pSA1. The cymbal as a structural acoustics demonstration: From evanescence to chaos. Kent L. Gee, Scott D. Sommerfeldt, Trevor A. Stout, Tracianne B. Neilsen, and Pegah Aslani (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

The cymbal can be used as a rich structural acoustics demonstration. Many well-defined normal modes spanning a broad range of frequencies can be evaluated using various techniques, including Chladni patterns, speckle interferometry, Doppler vibrometry, and acoustical holography. The literature shows modal patterns that are both membrane-like with nodal lines and circles as well as more complicated, irregular shapes. When the cymbal is struck forcefully, orthogonality gives way to modal coupling and resulting chaotic vibration that produces the characteristic shimmering sound. The sound radiation favors high-frequency modes; however, when the cymbal is put close to the ear, low frequencies dominate the sound heard. This low-frequency ringing persists well after the rest of the sound has decayed. This auralized evanescence will be demonstrated and further described as part of the presentation.

2:25

4pSA2. Demonstrations of fundamental topics in structural acoustics using a simply-supported plate. Andrew R. Barnard (Mech. Eng. - Eng. Mech., Michigan Technolog. Univ., 815 R.L. Smith MEEM Bldg., 1400 Townsend Dr., Houghton, MI 49931, arbarnar@mtu.edu) and Stephen A. Hambric (Graduate Program in Acoust., Penn State Univ., State College, PA)

As part of the ACS 519 course, sound-structure interaction, offered through the Penn State Graduate Program in Acoustics a set of demonstrations was developed to help reinforce structural acoustics theory. A simply-supported plate was used to develop a set of four concept demonstrations: mobility functions, radiated sound directivity, fluid loading, and acoustic transmission loss. A modal impact hammer, accelerometers, microphones, a sound level meter, and a sound intensity probe were used in combination with National

Instruments compact DAQ systems and LabVIEW software to develop these custom demonstrations. Basic theory and setup of the demonstrations are presented as well as videos of the demonstrations themselves.

2:45

4pSA3. A glass half full: Demonstrations of some surprising effects of fluid loading on the resonance of wine glasses (and other vessels). Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

A popular and (perhaps) provocative demonstration in an introductory course on acoustics and vibration involves asking students to predict what happens to the tap tone of a wine glass when liquid is added. Although their predictions are often incorrect, students readily accept the explanation of mass loading for why the pitch decreases, having already seen what happens to the frequency of a mass-spring system when mass is added. But is the pitch-lowering effect the same for all vibrational modes? Does it depend on the shape or thickness of the glass? For students beyond the introductory level, these questions stimulate deeper thinking about fluid-structure interaction. Predictions can be verified with a very simple apparatus.

3:05

4pSA4. Vibrational modes of a thin bar. Steven L. Garrett (Grad. Prog. in Acoust., Penn State, Appl. Res. Lab, P. O. Box 30, State College, PA 16804, sxg185@psu.edu)

Although most traditional acoustics textbooks that treat vibration (e.g., Rayleigh, Lamb, Morse, Morse and Ingard, Kinsler and Frey) address the longitudinal, torsional, and flexural modes of a thin bar, they do not stress the importance of sound speed measurements for the experimental determination of elastic moduli. This presentation will demonstrate a simple and inexpensive apparatus that uses two coils of copper wire and permanent magnets, developed by Bob Leonard and Izzy Rudnick at UCLA, that can be used to excite and detect all three modes of a free-free bar, accurately measure their modal frequencies, and use those results to provide accurate and precise values for the elastic moduli of the bar material [JASA **88**(1), 210 (1990)].

3:25

4pSA5. Standing waves on an electrically heated wire, revisited: Demonstration of glow at the nodes. Murray S. Korman and Ted McClanahan (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

In the experiment, a nichrome wire is suspended between a metal bridge and a pulley with a weight hanger and standard weights that supply a constant tension. The 28 gauge wire has a diam = 0.32 mm and a mass/length = 0.66 g/m. For a 2 m long wire with a 0.43 kg mass providing tension, the standing wave resonant frequencies are integer multiples of 20 Hz (neglecting the wire's stiffness). An AC current from a variable autotransformer is used to heat the wire (from the bridge to the pulley) until it glows. Near the bridge, a variable gap neodymium magnet is positioned with its poles on either side of the wire. The wire will glow at the nodes, but due to cooling (by the air vibration) the antinodes do not glow. The experiment is described in [C. A. Taylor, *The Art and Science of Lecture Demonstration*, Taylor & Francis, 1988]. Replace the magnet with an oscillator and audio power amplifier for frequency control [Sabatier and Drewrey, J. Acoust. Soc. Am. **104**, 1793 (1998)]. One can also demonstrate glowing nodes for "standing waves in a circle," by modifying the setup of [H. F. Meiners, Am. J. Phys. **33**, xiv, Oct. 1965].

3:45

4pSA6. Guitar pickups and the plucked string. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, drussell@engr.psu.edu)

This demonstration will use optical, magnetic, and piezoelectric guitar pickups to measure time signals and frequency spectra for the displacement, velocity, and force (measured at the bridge) of a plucked and struck guitar string. For the plucked string, experimental observations of the displacement and velocity of a point on the string agree very well with theory. Frequency spectra also agree well with theoretical predictions and highlight the differences in timbre between the audio signals obtained with each pickup. The demonstration will briefly discuss theoretical time signals and frequency spectra for a point on a plucked string, show measured time signals and frequency spectra from the various pickups, and compare audible examples of the sounds from the pickups for a plucked string. For the struck string, a simple theory approximates the basic shape of the string displacement and velocity, but dispersion effects in the string and the reality of the hammer-string interaction are evident in the measured string response. If time permits, a demonstration of the sound and spectrum of a compound string will be included.

Session 4pSC

Speech Communication: Indexical Factors in Speech Perception and Production (Poster Session)

Benjamin Munson, Chair

Speech-Language-Hearing Sciences, University of Minnesota, 115 Shevlin Hall, 164 Pillsbury Drive SE, Minneapolis, MN 55455

Posters will be on display from 1:00 p.m. to 5:00 p.m. To allow contributors an opportunity to view other posters, contributors of odd-numbered papers will be at their posters from 1:15 p.m. to 2:45 p.m., and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 4:30 p.m. There will be a 15-minute break from 2:45 p.m. to 3:00 p.m.

Contributed Papers

4pSC1. Acoustic effects of loud speech across vowels and speakers. Laura L. Koenig (Haskins Labs and Long Island Univ., 300 George St., New Haven, CT 06511, koenig@haskins.yale.edu) and Susanne Fuchs (Zentrum für Allgemeine Sprachwissenschaft [ZAS], Berlin, Germany)

Although loud speech is widely agreed mainly to involve increased respiratory driving support leading to greater acoustic intensity, past work suggests that speakers may make laryngeal and supralaryngeal adjustments as well. For example, previous studies suggest that loud speech is accompanied by greater jaw displacements as well as formant frequency variation, especially in the form of higher first formants. Acoustic studies have generally not sampled widely across the vowel space, however. This study assessed how formants, f0, duration, and intensity varied across conversational and loud conditions, with a particular focus on formant frequencies. Eleven German-speaking women performed three speaking tasks: Reading, answering questions, and recounting a recipe. Target words included a range of high, low, tense, and lax vowels. Loudness variation was elicited naturalistically via changing interlocutor distance. Initial results from the reading and question-answer tasks suggest that formant frequencies vary on average in louder speech, particularly for F1, but the effects are speaker-dependent and may also differ between high vs. low and tense vs. lax vowels. These acoustic data will ultimately be combined with simultaneously recorded data on intraoral pressure, vocal-fold contact, and breathing kinematics to assess the range of speakers' strategies for achieving louder speech.

4pSC2. Interactions among phonetic reduction and sociolinguistic variation in word recognition. Zack E. Jones and Cynthia G. Clopper (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, jones.5028@osu.edu)

A number of linguistic factors influence the phonetic realization of segments within a word, especially vowels within the word. These factors include lexical frequency, lexical neighborhood density, semantic predictability, and mention within a discourse. Sociolinguistic factors, such as speaking style and regional dialect, are also known to influence the phonetic realization of vowels. This study investigates the ways in which phonetic reduction caused by the linguistic factors and sociolinguistic variation from speaking style and talker dialect affect lexical processing. Participants are asked to identify single word tokens in noise that vary in frequency, density, predictability, discourse mention, speaking style, and talker dialect in a fully crossed design. We expect participants to identify words containing vowels with less phonetic reduction, a clear speaking style, and a familiar regional dialect with greater accuracy than words with more reduced vowels, a plain speaking style, or a less familiar dialect. The experimental design allows us to explore the individual contributions to word intelligibility of each of the linguistic and sociolinguistic factors independently and in complex interactions.

4pSC3. Talker normalization and acoustic properties both Influence spectral contrast effects in speech perception. Ashley Assgari and Christian Stilp (Univ. of Louisville, 317 Life Sci. Bldg., Louisville, KY 40292, ashley.assgari@louisville.edu)

Talker normalization (TN) occurs when listeners adjust to a talker's voice, resulting in faster and/or more accurate speech recognition. Several have suggested that TN contributes to spectral contrast, the perceptual magnification of changing acoustic properties. Studies using sine tones in place of speech demonstrated that talker information is not necessary to produce spectral contrast effects (Laing *et al.*, 2012 *Front. Psychol.*). However, sine tones lack acoustic complexity and ecological validity, failing to address whether TN influences spectral contrast in speech. Here we investigated how talker and acoustic variability influence contrast effects. Listeners heard sentences from a single talker (1 sentence), HINT (1 talker, 200 sentences) or TIMIT databases (200 talkers, 200 sentences) followed by the target vowel (varying from "ih" to "eh" in F_1). Low (100–400 Hz) or high (550–850 Hz) frequency regions were amplified (+20, +5 dB) to encourage "eh" or "ih" responses, respectively. When sentences contained +20 dB spectral peaks, contrast effect magnitudes were comparable across conditions. When sentences contained +5 dB peaks, contrast effect magnitudes decreased overall, but were smallest following TIMIT sentences with larger, comparable effects for single-talker conditions. Thus, TN influences contrast effects when spectral peaks are modest, but not when they are large.

4pSC4. Speaker age effects on the voicing contrast of Tokyo Japanese stops. Zack E. Jones and Chris S. Rourke (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, jones.5028@osu.edu)

While older Japanese speakers from the Kansai region produce voiced-voiceless stop contrasts with a true voicing distinction, older speakers from the northern Tohoku region produce the same stop contrasts with a short- versus long-lag VOT distinction, similar to the voicing contrast of English stops. However, Japanese speakers of younger generations from both of these dialect regions have been observed to produce the voiced-voiceless stop contrast in much the same way to each other, and they also seem to be using VOT as a less informative cue for stop distinction than speakers of previous generations (Takada, 2011). This study investigates the status of this sound change in the speech from the Tokyo dialect region, generally considered the standard for the modern language. Using speech data available from the Corpus of Spontaneous Japanese, we analyze stops produced by Tokyo speakers to determine differences in voicing contrast conditioned by speaker age and gender. We expect the stop contrast of older speakers to pattern differently than that of younger speakers, and we expect to see evidence for a sound change in progress similar to that observed in the other dialects. We also explore other acoustic correlates that may potentially contribute to Japanese stop contrasts as the sound change progresses.

4pSC5. Regional variation of vowels in Saterland Frisian. Heike Schoormann, Wilbert Heeringa, and Jörg Peters (Inst. of German Studies, Carl von Ossietzky Univ. Oldenburg, Ammerlaender Heerstraße 114-118, Oldenburg 26129, Germany, heike.schoormann@uni-oldenburg.de)

Saterland Frisian is spoken in three neighboring villages in North-West Germany, Strücklingen, Scharrel, and Ramsloh. In this study, we examine whether there is regional variation between the vowel systems of the three varieties. Speakers were instructed to read monophthongs and diphthongs in a neutral /hVt/ frame. Acoustic measurements included vowel duration, mid-vowel F1 and F2, the amount of Vowel Inherent Spectral Change (VISC, Nearey and Assmann, 1986), and the spectral rate of change (cf. Fox and Jacewicz, 2009*)*. Results confirm large inventories for the three varieties of Saterland Frisian, although some vowels have undergone a merger with neighboring categories. The comparison of single vowel categories in the three varieties did neither reveal variation of vowel duration nor of dynamic spectral properties. Regarding static spectral properties, however, dialectal variation was observable. In Scharrel, monophthongs are more centralized in the F1 dimension. This finding is discussed with respect to the common view of local people that speakers from Scharrel speak faster than speakers from the other two places.

4pSC6. Vowels of Korean dialects. Yoonjung Kang (Dept. of Linguist, Univ. of Toronto, 1265 Military Trail, Humanities Wing, H427, Toronto, Ontario M1C 1A4, Canada, yoonjung.kang@utoronto.ca), Jessamyn L. Schertz (Ctr. for French and Linguist, Univ. of Toronto Scarborough, Toronto, Ontario, Canada), and Sungwoo Han (Korean Lang. and Lit., Inha Univ., Seoul, South Korea)

This study compares the monophthongal vowels /a ɛ e i ʌ ɨ o u/ of two North Korean dialects as spoken by ethnic Koreans in China (24 Phyeongan and 21 Hamkyoung) with the vowels of Seoul Korean (25 younger and 32 older). Younger and older speakers of Seoul Korean are compared to examine the sound change in progress in Seoul. The most striking difference among the dialects is in the realization of /o/ and /ʌ/. In Seoul, /o/ is produced higher than /ʌ/. In Phyeongan, /o/ is lower than /ʌ/, while in Hamkyoung, the two are comparable in height and the main contrast is along F2. Also, /e/-/ɛ/ contrast is lost in Seoul but robust in the Northern dialects. Within Seoul Korean, the back vowel shift observed in recent literature is confirmed (Cho S. 2003, Han J. and Kang H. 2013, and Kang Y. to appear)—/o/ is raised toward /u/ while /ɨ/ is fronted away from /u/ in younger speakers' speech. In contrast to recent reports of /u/-/ɨ/ and /o/-/ʌ/ merger in homeland North Korean dialects (Kang S. 1996, 1997, Kwak 2003, and So 2010), in our Northern data, these contrasts remain distinct.

4pSC7. Diachronic change in perception of Korean sibilants. Jessamyn L. Schertz, Yoonjung Kang (Ctr. for French and Linguist, Univ. of Toronto Scarborough, 1265 Military Trail, Humanities Wing, HW427, Toronto, Ontario M1C 1A4, Canada, jessamyn.schertz@utoronto.ca), Sungwoo Han (Korean Lang. and Lit., Inha Univ., Seoul, South Korea), and Eunjong Kong (Dept. of English, Korea Aerosp. Univ., Seoul, South Korea)

The laryngeal contrast in the Seoul dialect of Korean is in a state of flux: the increasing importance of f0 (relative to VOT) in both perception and production of the three-way stop contrast in younger (vs. older) Seoul speakers has been well-documented. The current work turns to perception of the Korean sibilant series, comprised of a three-way affricate contrast, (fortis vs. lenis vs. aspirated, parallel to the stop contrast) and a phonologically ambiguous two-way fricative contrast (fortis vs. "nonfortis"). We map younger (mean 33 years old) and older (mean 66) Seoul listeners' perceptual spaces for the sibilant class using a five-way forced-choice task, with stimuli manipulated to vary independently across multiple acoustic dimensions (consonantal spectral information, vocalic spectral information, frication duration, aspiration duration, and f0). Hierarchical classification tree analyses reveal systematic age-related differences in cue-weighting. While both age groups make use of a combination of spectral properties of both the consonant and vowel, temporal information, and f0 when categorizing the stimuli, f0 plays a greater role in predicting sibilant classification in younger as compared to older listeners. Furthermore, categorization patterns suggest that the sound change not only affects perception of the three-way laryngeal contrast, but also has implications for other phonological contrasts (e.g. affricate vs. fricative "manner" contrast).

4pSC8. The southern vowel shift in women from Mississippi. Whitney Knight and Wendy Herd (English, MS State Univ., 2004 Lee Hall, MS State, MS 39762, whitneyleighknight@gmail.com)

Though previous research has documented the Southern Vowel Shift (SVS) in Alabama and Tennessee, no research has focused on the SVS in Mississippi. The majority of SVS research has also focused on European-Americans and assumed that African-Americans do not participate in the shift. The SVS consists of three stages: /aɪ/ monophthongization; lowering and centralizing of /e/ toward /ɛ/ and raising and peripheralizing of /ɛ/ toward /e/; and lowering and centalizing of /i/ toward /ɪ/ and raising and peripheralizing of /ɪ/ toward /i/. In this study, data were collected from women from northern (N = 11) and central (N = 23) Mississippi, with central residents evenly recruited from urban and rural areas. Of these, 15 were European-American and 19 were African-American. Participants read a list of words including the target vowels in b_t and b_d frames, and then F1 and F2 were measured at five equidistant points. F1, F2, and trajectory length were analyzed to determine to what extent participants exhibited the SVS. There were effects of Region, Rurality, and Race such that central residents shifted more than northern residents, rural residents shifted more than urban residents, and African-American residents shifted more than European-American residents. These results suggest that African-Americans do participate in the SVS.

4pSC9. /l/-darkness in Newfoundland English. Sara Mackenzie, Paul De Decker, and Rosanna Pierson (Linguist, Memorial Univ. of NF, Sci. Bldg. 3050B, St. John's, New Foundland A1B 3X9, Canada, pauldd@mun.ca)

This paper reports on the first acoustic study on the allophonic distribution of /l/ in Newfoundland, Canada, where Irish-settled varieties of English exhibit light variants in both coda and onset positions (Clarke 2010, Paddock 1982). This pattern is distinct from standard North American English which has a dark, or velarized form in coda position and light [l] in onsets (Halle & Mohanan 1985). Productions from 10 male and 13 female speakers from across the province were collected and spectrally analyzed using Praat 5.4 (Boersma & Weenink 2014). Darkness measurements (F2-F1 at the midpoint of each /l/) were calculated and within-subjects t-tests conducted to examine differences across onset and coda positions. Results show the standard North American pattern exists within our sample, with significantly darker /l/s in coda position. However, some speakers showed coda darkness values comparable to onset /l/s of those speakers with the allophonic variation. Others had darkness values comparable to coda realizations in all positions. These findings are discussed using a model of dialect contact that identifies how, despite the presence of the dominant North American pattern, some speakers of Newfoundland English maintain a several hundreds year old pattern which lacks allophonic variation across syllable positions.

4pSC10. Production and perception among three competing pre-/l/ mergers. Lacey R. Arnold (English, North Carolina State Univ., 1214 Carlton Ave., Raleigh, NC 27606, lrarnold@ncsu.edu)

This study examines the status of three patterns of merger among /ul/, /ol/, and /ʊl/ in Youngstown, Ohio. Using acoustic analyses of F1 and F2 and multiple-forced-choice perception task results from 40 Youngstown natives ages 9–81, this study addresses the progression of these mergers in apparent time, the perceptual correlates of merged and distinct production, and maintenance of durational distinctions in production and/or perception. Initial analysis of perception data suggests that different patterns of merger are progressing differently in the community and that production does not directly correlate with perception, perhaps as a result of exposure to multiple patterns of merger in the community. Despite the different patterns of merger in the community, speakers seem to pick up on durational cues in perception tasks.

4pSC11. Dialectal duration variations reveal historical sound change. Di Wu and Chilin Shih (Linguist, Univ. of Illinois at Urbana-Champaign, 4080 Foreign Lang. Bldg., 707 S Mathews Ave., Urbana, IL 61801, diwu4@illinois.edu)

This study models duration patterns of different dialectal regions to evaluate the spread of a phonological process. Er [ɚ] is a diminutive suffix in Chinese and is a major linguistic feature differentiating northern and southern Mandarin speech. This study evaluates the duration of the er-suffix along with its host syllable in a sentence reading task. The participants were from three regional groups chosen to examine the spread of this feature—Beijing, Mid-China, and Taiwan. Linear regression models were used to analyze duration variations quantitatively. Results reveal that Beijing speakers exhibit the most advanced er-suffixation, as expected. Duration values of words with or without the er-suffix are nearly identical, suggesting that er has been incorporated into the host syllable and has lost its own syllabic status. In contrast, speakers from Taiwan show the least advanced er-suffixation, where the suffix maintains its syllabic status with full duration. Speakers from Mid-China show an intermediate stage in accord with the geographical location. The duration contrast is consistent with reported diachronic phonological changes. The method suggests that duration patterns can be used to evaluate the spread of phonological processes, projecting synchronic, regional variations to their diachronic development.

4pSC12. Talker variation in the perception of speech in noise. Noah H. Silbert and Lina Motlagh Zadeh (Commun. Sci. & Disord., Univ. of Cincinnati, 3202 Eden Ave., 344 French East Bldg., Cincinnati, OH 45267, noah.silbert@uc.edu)

Speech communication commonly occurs in the presence of noise. Research on the perception of speech in noise has largely focused on the perceptual availability of target phonetic information in the presence of various noise maskers (e.g., white noise, speech-shaped noise, temporally modulated noise, and multi-talker babble). Previous work has shown that a glimpsing model and a modified Articular Index model closely approximate overall accuracy of noise-masked speech perception (Cooke, 2006, *JASA*; Allen, 2005; *JASA*). Some recent work has focused on variation in confusion patterns across listeners (Silbert, 2012, *JASA*, 2014, *Lab. Phon.*) and on difference in perceptual error rates across and within consonant categories (Toscano & Allen, forthcoming, *JSLHR*). The present work focuses on listener and stimulus-talker variation in the identification of consonants. Eleven listeners identified numerous tokens of each of four consonants (t, d, s, z) in CV syllables produced by 20 talkers (10 male, 10 female) masked by 10-talker babble. Data plots and multilevel logistic regression model fits indicate substantial variation in talker-specific influences on identification accuracy as well as substantial interactions between talker and stimulus category. Analysis of perceptual confusion patterns via a multilevel, multidimensional Gaussian signal detection model will also be presented.

4pSC13. Human recognition of familiar voices. Stanley Wenndt (Air Force Res. Lab., 701 N George St., Rome, NY 13440, swenndt@twcny.rr.com)

Recognizing familiar voices such as friends and co-workers is something we do every day. In typical environments such as an office or home, it is usually easy to identify who is speaking. In noisier environments, this can become a difficult task. This research examines how robust listeners are at identifying familiar voices in noisy, changing environments and what factors may affect their recognition rates. While there is previous research addressing familiar speaker recognition, it is a difficult topic to research since the focus is on familiar voices. The data being used for this research was collected in such a fashion to mimic conversation, free-flow dialog, but in a way to eliminate many variables such as word choice, intonation or non-verbal cues. This data provides some of the most realistic test scenario to-date for familiar speaker identification. A pure-tone hearing test was used to separate speakers into normal hearing and hearing impaired groups. It is hypothesized that the results of the Normal Hearing Group will be statistically better than then results of the Hearing Impaired Group. Additionally, a new aspect of familiar speaker recognition is addressed by having each listener rate his or her familiarity with each speaker.

4pSC14. Sensitivity to the emotional tone of verbal commands: Executive-function and chemotherapy effects. Blas Espinoza-Varas, Sudha Lakhwani (Commun. Sci. & Disord., OU Health Sci. Ctr., 1200 N. Stonewall Ave., Oklahoma City, OK 73117-1215, blas-espinoza-varas@ouhsc.edu), Ashan Khan (IPS Researrch, Oklahoma City, OK), and Kai Ding (Biostatistice & Epidemiology, OU Health Sci. Ctr., Oklahoma City, OK)

Treatment adherence requires complying with verbal commands ("quit smoking!") issued to impede or instigate behaviors, and a lenient or stern emotional voice tone (VT) calls for optional or mandatory adherence. The ability to identify the commands VT in conditions imposing executive-function (EF) demands could estimate the likelihood of achieving adherence. Prior to chemotherapy and between cycles 3–4 of gynecological cancer, we assessed VT identification under three EF demands. 1) Inhibitory control trials presented the cue word "left" or "right" followed by impeding commands in lenient or stern tone, mapped onto a left or right response; the cue and ear side could be congruent or in conflict with the correct response side. Trials presenting instigating commands ("go!") mapped lenient or stern onto a right or left response. 2) Response-mapping switching conditions interleaved impeding and instigating commands within the same trial block, and required switching the mapping rule depending on the command, impeding or instigating. 3) Working-memory conditions asked whether the command presented on the current trial was equal to or different from the one presented two trials back. Without EF demands, VT identification errors were few, but increased significantly with EF demands, being largest in condition 2; chemotherapy effects were small.

4pSC15. A language familiarity effect for talker identification in forward but not time-reversed speech. Sara C. Dougherty, Deirdre E. Mclaughlin, and Tyler K. Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 653 Commonwealth Ave., Boston, MA 02215, sarad12@bu.edu)

Listeners identify voices more accurately in their native language than an unfamiliar foreign language—a phenomenon known as the Language Familiarity Effect (LFE). The purpose of this study was to assess two hypotheses about the source of the LFE: 1) that the LFE depends on linguistic processing of speech, or 2) that it results from listeners' familiarity with language-specific but non-linguistic properties of speech acoustics. We trained native speakers of English (N = 16) and Mandarin (N = 16) to identify English and Mandarin voices. Stimuli consisted of unaltered recordings of ten sentences in those languages, as well as time-reversed versions of the same. Importantly, time-reversed speech preserves global acoustic properties of the stimuli while rendering them completely incomprehensible. Consistent with the LFE, participants identified voices in their native-language significantly more accurately than in the foreign language when listening to normal, forward speech. However, participants did not exhibit a corresponding LFE for the time-reversed speech stimuli: time-reversed native- and foreign-language voices were identified with equal accuracy. These findings support the hypothesis that processes involved in speech comprehension are the source of the LFE rather than differences in the non-linguistic analysis of speech acoustics.

4pSC16. Effects of reading ability on native and nonnative talker recognition. Minal Kadam (Univ. of Connecticut, Storrs, CT), Adriel J. Orena (McGill Univ., Montreal, Quebec, Canada), Rachel M. Theodore (Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu), and Linda Polka (McGill Univ., Montreal, Quebec, Canada)

Recent findings suggest that phonological knowledge influences talker identification. Specifically, talker identification is improved for native compared to nonnative talkers, and adults with reading disability show impaired talker identification even for native talkers [Perrachione *et al.*, Science, **333,** 595 (2011)]. Here, we examine whether effects of reading ability on talker identification emerge among unimpaired readers. Monolingual English adults were assigned to either the high or low reading group based on standardized assessments of reading and reading sub-skills. All readers learned to identify the voices of four English talkers and four French talkers. Training consisted of a two-alternative forced choice task with feedback provided on every trial. After training, retention of learning was tested using a four-

alternative forced choice task without feedback. The results to date suggest that the high reading group learned both the native and nonnative voices faster compared to the low reading group. Moreover, the high reading group showed increased retention of learning compared to the low reading group, but only for the nonnative voices. These results are consistent with recent findings demonstrating an effect of language proficiency on talker identification, and extend them to include a gradient role for native language phonological ability on nonnative talker identification.

4pSC17. Cross-modal transfer of talker learning. Dominique C. Simmons, James W. Dias, Josh Dorsi, and Lawrence D. Rosenblum (Psych., Univ. of California Riverside, 900 University Ave., Riverside, CA 92521, dsimm002@ucr.edu)

Observers can match unfamiliar faces to corresponding voices (Kamachi *et al*., 2003). Observers can also use dynamic point-light displays containing isolated visible articulations to match faces to voices (Rosenblum *et al*., 2006) and to sinewave versions of those voices (Lachs & Pisoni, 2004) suggesting that isolated idiolectic information can support this skill. Cross-modal skills also extend to facilitation of speech perception. Familiarity with a talker in one modality can facilitate speech perception in another modality (Rosenblum, Miller, & Sanchez, 2007; Sanchez, Dias, & Rosenblum, 2013). Using point-light and sinewave techniques, we tested whether *talker* learning transfers across modalities. If learning of idiolectic talker information can transfer across modalities, observers should better learn to auditorily recognize talkers they have previously seen. Sixteen subjects trained to recognize five point-light talkers. Eight of these subjects then trained to recognize sinewave voices of the five previously seen talkers (Sheffert *et al*., 2002). The remaining subjects trained to recognize five sinewave voices of new talkers. Subjects trained to recognize voices of talkers who they had previously seen performed better than subjects trained to learn voices of new talkers, $t(14)=-1.834$, $p<0.05$. Results suggest that learning idiolectic talker-specific information can transfer across modalities to facilitate talker learning.

4pSC18. Exposure to an unfamiliar language bolsters talker learning. Adriel John Orena (School of Commun. Sci. and Disord., McGill Univ., 2001 McGill College, 8th Fl., Montréal, Quebec H3A 1G1, Canada, adriel.orena@mail.mcgill.ca), Rachel M. Theodore (Dept. of Speech, Lang., and Hearing Sci., Univ. of Connecticut, Storrs, CT), and Linda Polka (School of Commun. Sci. and Disord., McGill Univ., Montréal, Quebec, Canada)

Listeners are better at identifying talkers who speak their native language than talkers who speak a foreign language, showing that phonological knowledge of a language facilitates talker identification. However, research with infants indicates that language comprehension is not necessary for improving talker identification. In this study, we asked whether language exposure alone could improve talker learning. Two groups of English-monolingual adults were recruited: one group from Montréal, Québec, who receive regular French exposure, and the other from Storrs, Connecticut who receive no French exposure. In Experiment 1, we used a four-alternative forced choice task (4AFC) to train listeners about the voices of four English talkers and four French talkers. Results show that Montréal participants were faster at learning French voices than Storrs participants, showing that exposure to a foreign language is sufficient in boosting talker learning in that language. However, in Experiment 2, a 2AFC was used to train participants, and no significant differences were found between groups. These findings show that varying the training paradigm in laboratory analogs of talker identification can induce different types of talker learning. Taken together, our results suggest that phonological sensitivity contributes to listeners' talker identification abilities, but only under certain training contexts.

4pSC19. Processing talker variability in semantic/associative priming: Does talker voice matter? Yu Zhang and Chao-Yang Lee (Ohio Univ., Grover Ctr. W239, Athens, OH 45701, yz137808@ohio.edu)

The effect of talker variability on lexical access is investigated using short-term semantic/associative priming experiments. Prime-target pairs, either semantically associated (e.g., king-queen) or unrelated (e.g., bell-queen), were spoken by the same or different male speakers. Two interstimulus intervals (ISI, 50 and 250 ms) were used to explore the time course of semantic/associative priming and voice specificity effects. Forty listeners completed a lexical decision task followed by a talker voice discrimination task on the same auditory stimuli. Results from the lexical decision task showed semantic/associative priming effects, although the magnitude of priming was unaffected by talker variability or ISI. Results from the talker voice discrimination task showed no priming effects, although different-talker trials elicited faster and more accurate responses. In comparison with previous results using a similar paradigm (Lee & Zhang, in press; Zhang & Lee, 2011), this set of data suggests that talker variability might not influence access to semantic aspects of spoken language. Furthermore, voice discrimination task generated different patterns (e.g., lack of priming) from the lexical decision task. This suggests that the extent of talker voice effect on lexical access may be subject to attention manipulation.

4pSC20. Gender and age differences in vowel-related formant patterns: What happens if men, women, and children produce vowels on different *and* on similar F0? Dieter Maurer, Heidy Suter (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Toni-Areal, Pfingstweidstrasse 96, Zurich 8031, Switzerland, dieter.maurer@zhdk.ch), Daniel Friedrichs, and Volker Dellwo (Dept. of Comparative Linguist, Univ. of Zurich, Zurich, Switzerland)

There is a broad consensus in the literature that vowel-specific formant patterns differ as a function of gender (men/women) or age (adults/children) due to different average vocal tract sizes. Although an additional influence of fundamental frequency F0 is discussed in corresponding normalization approaches, formant patterns relating to sounds of adults and children that exhibit the same F0, to sounds of adults with higher F0 than sounds of children, and to sounds of men with higher F0 than sounds of women are barely compared. Investigating vowels of men, women, and children producing sounds with varying F0, we observed (1) a possible decrease or even a disappearance of the expected speaker-group differences in the formant frequencies $<$ 1.5 kHz if F0 of the utterances correspond for children, women, and men, and (2) a possible "inversion" of the expected speaker-group differences $<$ 1.5 kHz if F0 of the utterances of adults are higher than those of children, or F0 of men are higher than those of women. However, no corresponding relationship between F0 and the higher formants $>$ 1.5 kHz was found. These observations call for a further examination of the role of F0 when interpreting speaker-group related differences in formant patterns.

4pSC21. Speaker sex discrimination performance for voiced and whispered vowels at very short durations. David R. Smith (Psych., Univ. of Hull, Cottingham Rd., Hull HU6 7RX, United Kingdom, d.r.smith@hull.ac.uk)

When listening to someone's voice what duration of stimulus is required to tell whether the person speaking is a man or a woman? Previous research studied this using *voiced* speech [D. R. R. Smith, Acta Psychologica **148**, 81–90 (2014)]. The current study expanded this analysis by investigating what duration of stimulus is required to discriminate speaker sex when listening to *whispered* speech. Psychometric functions were collected plotting percent correct discrimination that a man or woman spoke, as a function of very brief vowel segment durations (up to 60 ms), for both voiced and whispered vowels. Results show that speaker sex discrimination performance is significantly impaired for whispered vowels, as compared to voiced vowels, for all durations tested. These findings are interpreted in terms of the impoverished cues to speaker sex in whispered compared to voiced speech.

4pSC22. Gender normalization in fricative perception in single- and mixed-gender blocks. Benjamin Munson (Univ. of Minnesota, 115 Shevlin Hall, 164 Pillsbury Dr., SE, Minneapolis, MN 55455, benjamin.ray.munson.jr@gmail.com)

Previous research has shown that listeners' identification of English anterior sibilant fricatives changes depending on whether they are primed to believe that the talker is a woman or a man, by pairing audio stimuli with images or videos of a woman or a man (Munson, 2011; Strand & Johnson, 1996; Winn *et al*., 2013). The current experiment is part of a program of research that endeavors to understand the nature of imputed gender effects in speech perception. Previous studies used fully within-subjects designs: listeners were presented with all stimuli combined with both male and female faces. The current experiment examined whether gender normalization occurs equally strongly in within-groups designs, and between-groups designs in which listeners are presented with only male or female faces. Four groups of listeners identified a nine-step *sack-shack* continuum created by combining a s-ʃ continuum with a natural production of a VC that had been acoustically manipulated to be gender-neutral. Listeners were presented with a male face only, a female face only, both male and female faces in separate trials, or no face. Results will indicate how robust gender normalization effects are across different experimental conditions.

4pSC23. Identification of talker gender and speech production mode from high-pass filtered vowel segments. Jeremy Donai (Commun. Sci. and Disord., West Virginia Univ., 3601 4th St., Lubbock, TX 79430, jeremy.donai@ttuhsc.edu) and Dwayne Paschall (Speech-Lang-Hearing Sci., Texas Tech Univ. Health Sci. Ctr., Lubbock, TX)

This study reports gender and production mode judgments from high-pass filtered vowel segments from 22 listeners with normal-hearing. Two adult males and two adult females produced vowels in an h/Vowel/d (hVd) context in both a spoken and sung mode of production. The hVd utterances were produced using the carrier phrase, "I say (insert hVd) again." The speakers produced the sung utterances to the tune of "Old McDonald." The vowels included /i/ as in "heed", /ɝ/ as in "heard", and /ɔ/ as in "hawd." A 250 millisecond segment from the central portion was extracted and windowed. The signals were then high-pass filtered at 3.2 kHz to remove low-frequency spectral detail. The listeners were positioned in front of a graphical user interface and instructed to click on one of four buttons, "Male spoken," "Male sung," "Female spoken," or "Female sung" to indicate the perceived gender and mode of production. Results showed below chance performance for the male sung and female spoken utterances and performance significantly above chance for the male spoken and female sung utterance. Signals with similar unfiltered F0 values (i.e., male sung and female spoken) were discriminated compared to those with the highest (female sung) and lowest (male spoken) F0s.

4pSC24. Vowels or consonants: Which is more effective in distinguishing between self-identified gay and heterosexual male speakers of American English? Erik C. Tracy (Psych., Univ. of North Carolina Pembroke, PO Box 1510, Pembroke, NC 28372, erik.tracy@uncp.edu)

Previous research (Tracy & Satariano, 2011) investigated how listeners were able to distinguish between self-identified gay and heterosexual male speakers of American English. In one experiment, listeners heard greater and greater portions of utterances that followed a CVC pattern (/m/, /mæ/, and /mæs/), and their judgments significantly improved upon hearing the second phone. It was unclear if this result was due to the second phone being a vowel or because it was an additional phone. It was hypothesized that judgments would improve if listeners heard additional phones. Furthermore, listeners would primarily rely on vowels, and not consonants, to form their judgments. This hypothesis was tested in the present experiment. Listeners heard utterances that contained one, two, or three consonants, and one, two or three vowels. The results demonstrated that sexual orientation judgments improved if the utterances contained three phones compared to one phone. It was also discovered that the listeners were better able to distinguish between the gay and heterosexual speakers if the utterances contained vowels rather than consonants. Thus, the hypotheses were confirmed. While sexual orientation judgments improved if the utterances contained more phones, listeners were better able to distinguish between the speakers if the utterances contained vowels.

4pSC25. The role of social information in cognitive processing: Sex and sexuality. Eric Wilbanks (English, North Carolina State Univ., 204 Tompkins Hall, Campus Box 8105, Raleigh, NC 27695, ewwilban@ncsu.edu)

Exemplar models of cognitive linguistic processing hold that humans' robust memory faculty allows for the construction of "exemplars" or prototypes gained through statistical procedures applied to experiential memories of variation (Pierrehumbert, 2006). Since they are based upon experiential memories, exemplar clouds are sensitive to social information. As Mendoza-Denton *et al.* (2003) note, congruence between the variant and the center of the exemplar cloud often facilitates cognitive processing. Both speaker age (Walker and Hay, 2011) and speaker sex (Sumner and King, 2013) have been shown to affect semantic processing. Expanding upon these investigations, the current study examines a new social variable, sexuality. First, a lexical association task was carried out to construct a corpus of word-pairs whose semantic links differed for straight female, male, and gay male speakers. Then, the effect of congruence between speaker sexuality, gender, and semantic pair was investigated in a lexical decision task. Statistical analyses of reaction times illustrate significant increases in processing when congruence between speaker and associated pair is negative. Additionally, the effect of (in)congruence on semantic priming was greatest for the gay male speaker. These data demonstrate the saliency of sexuality information in lexical processing and semantic linking and argue for an expansion of the social categories relevant for Exemplar models of linguistic processing.

4pSC26. Phonetic convergence during conversational interaction and speech shadowing. Jennifer Pardo, Adelya Urmanche, Sherilyn Wilman, Jaclyn Wiener, Hannah Gash, Sara Parker, Keagan Francis, and Alexa Decker (Psych., Montclair State Univ., 1 Normal Ave., Montclair, NJ 07043, pardo@optonline.net)

Phonetic convergence has been studied in both speech shadowing tasks and in conversational interaction. In both settings, phonetic convergence has been found to be highly variable, with higher convergence measures usually found in studies that used speech shadowing. In order to examine whether phonetic convergence in both settings arises from similar mechanisms, the current study compared phonetic convergence in both speech shadowing and in paired conversational interaction in the same set of talkers. A set of 96 talkers (48 female) provided shadowed recordings and participated in a paired conversational map task. Moreover, the pairs were constructed to permit comparisons of same- and mixed-sex pairings. Phonetic convergence was assessed in both tasks using AXB perceptual listening tests with naïve listeners. Overall, phonetic convergence was highly variable across pairings in both shadowing and conversational tasks, with interesting effects of talker and pair sex.

4pSC27. Phonetic convergence in multiple features. Chelsea Sanker (Linguist, Cornell Univ., 203 Morrill Hall, Ithaca, NY 14853, cas443@cornell.edu)

People's speech shifts during conversation and other interactions to become more similar to the speech of interlocutors in several linguistic characteristics. Most studies on this convergent shift analyze one or two phonetic features (e.g., Babel 2009, Nielsen 2011) or holistic perceptual similarity ratings (e.g., Goldinger 1998). This study focuses on analyzing how the amount of change in difference between speakers' averages in one feature is correlated with that pair's change in difference in other features. Convergence was compared for eight speech features over the course of partners' interactions: F1, F2, vowel duration, pitch, amplitude, turn duration, duration of pauses within turns, and duration of pauses between turns. Among the 28 correlations between convergence in different features, there were only three correlations which reached significance: between within-turn pause duration and between-turn pause duration, between pitch and F2, and between pitch and turn duration, all of which can be attributed to feature correlation independent of change. The degree to which convergence is exhibited by a pair in each feature seems to be highly influenced by the individual speakers; there was a significant correlation in the convergent change within each feature between pairs including the same individual.

4pSC28. Source versus spectral cues in the perception of indexical features in speech. Ewa Jacewicz, Robert A. Fox, and Hannah Ortega (Dept. and Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., 110 Pressey Hall, Columbus, OH 43210, jacewicz.1@osu.edu)

Spoken language includes two different forms of information: linguistic (message related) and indexical (related to individual speaker characteristics). This study explores the nature of the acoustic cues that listeners may use to identify gender and dialect. Spontaneous utterances were produced by 40 speakers (20 male, 20 female) from two different regional dialects spoken in central Ohio (OH) and western North Carolina (NC). These utterances were equally divided into three sets of tokens. One set was unprocessed (except for amplitude equalization). A second set was low-pass filtered at 400 Hz, retaining prosodic information, but little content. A third set was processed through an 8-channel noise vocoder eliminating all harmonic information (simulating cochlear implant processing). These three stimulus sets (blocked and randomized) were played to 20 OH listeners who identified whether the token was produced by a man or a woman, from OH or NC. Gender identification rates were high (means $>$ 89%) across all three conditions with clear $>$ LP filtered $>$ vocoded. The rates for dialect identification were significantly lower overall with the LP-filtered condition close to chance (58%). Discussion will center on listener use of acoustic features and perceptual sensitivity (d') to gender and dialect.

4pSC29. Talker variation and systematicity in voice onset time: A corpus study. Eleanor Chodroff and Colin Wilson (Cognit. Sci., Johns Hopkins Univ., Krieger Hall 237, 3400 N. Charles St., Baltimore, MD 21218, chodroff@cogsci.jhu.edu)

Previous research has demonstrated variation across talkers in the phonetic realization of speech sounds, including vowels (e.g., Peterson and Barney, 1952), fricatives (e.g., Newman *et al.*, 2001) and stop consonants (e.g., Allen *et al.*, 2003; Theodore *et al.*, 2009). The challenge that talker variation presents to perceptual processes may be significantly reduced if different aspects of talker-specific phonetics are strongly correlated. The present study employs a subset of read speech from the Mixer-6 corpus (Brandschain *et al.*, 2013), containing recordings from 129 native English participants (60 male). The voice onset time (VOT) of prevocalic word-initial stops (N = 59,075) was measured using forced alignment (Yuan & Liberman, 2008) and subphonemic postprocessing (AutoVOT; Sonderegger & Keshet, 2012). Talker means of the voiceless stops are highly correlated with each other (rs = /p/-/t/: 0.76; /p/-/k/: 0.79; /t/-/k/: 0.74; $ps < 0.0006$, alpha-corrected), as are /b/ and /g/ among the voiced stops ($r = 0.49$, $p < 0.0006$). Additionally, talker means of each stop category are correlated with corresponding standard deviations ($r = 0.87$, $p < 0.0001$). These suggest a degree of systematicity in variation, with implications for talker adaptation, as listeners may extrapolate from few input values to approximate talker-specific variance and cross-category means.

4pSC30. Examining correlations between phonetic parameters: Implications for forensic speaker comparison. Erica Gold (Linguist and Modern Lang., Univ. of Huddersfield, Heslington, York YO10 5DD, United Kingdom, erica.gold@york.ac.uk) and Vincent Hughes (Lang. and Linguistic Sci., The Univ. of York, York, United Kingdom)

The research presented in this paper builds upon a previous pilot study (Gold and Hughes 2012). This paper explores the correlation structure of speech parameters from a sociolinguistically homogeneous set of male speakers of Southern Standard British English using a series of segmental, suprasegmental and linguistic parameters. Data was extracted from a subset of speakers from the Dynamic Variability in Speech (DyViS) database (Nolan *et al.*, 2009) and consist of: midpoint F1, F2 & F3 values for /a ɔ u/, midpoint F1, F2 & F3 values hesitation markers UM and UH, dynamic F1, F2 & F3 values for PRICE /aɪ/, long-term formant distributions (LTFD) F1-F4, mean and standard deviation of fundamental frequency (F0), mean articulation rate (AR), voice onset time (VOT) for word-initial /t/ and /k/, and click rate (ingressive velaric stops). The results of the study present a complex correlation structure between linguistic-phonetic variables, and not all correlations are predicted by phonetic theory. The results of the correlations are discussed in relation to implications that exist when combining parameters for forensic speaker comparison casework; specifically, the caution that needs to be yielded by experts in casework to avoid over- or under-estimating the strength of evidence.

4pSC31. The relationship between fundamental frequency and within-speaker vowel reduction. Christina Kuo (Commun. Sci. and Disord., James Madison Univ., MSC4304, 801 Carrier Dr., Harrisonburg, VA 22807, kuocx@jmu.edu), Gary Weismer (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Madison, WI), and Casey Behre (Commun. Sci. and Disord., James Madison Univ., Harrisonburg, VA)

The contribution(s) of fundamental frequency (F0) to vowel production has potential theoretical and clinical implications. The sufficient contrast hypothesis [Diehl *et al.*, J. Phon. **24**, 187–208 (1996)] suggests that a high F0 is associated with effects of spectral undersampling, and thus may require more exaggerated formant frequencies to maintain the perceptual distinctiveness of vowels. Nonetheless, studies on the interplay between F0 and vowel acoustics would appear inconclusive [Byrd, J. Acoust. Soc. Am. **92**, 593–596 (1992)] [Weirich & Simpson, J. Acoust. Soc. Am. **134**, 2965–2974 (2013)]. This study evaluated the sufficient contrast hypothesis as a within-speaker mechanism, with an overarching hypothesis that formant frequency exaggeration associated with spectral undersampling, if any, occurs within a given production system (i.e., speaker). To this end, formant frequencies of vowels and F0 were obtained for the same speaker across different speaking tasks. Within speaker, it was hypothesized that vowel formant frequencies as well as F0 change across tasks. More importantly, average F0 is hypothesized to be negatively correlated with the degree of vowel formant frequency reduction within-speaker across tasks. That is, a vowel system produced with a higher F0 would be less prone to reduction because the spectral peaks are better defined.

4pSC32. Within- and between-talker variability in voice quality in normal speaking situations. Jody Kreiman (Head and Neck Surgery, UCLA, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90095, jkreiman@ucla.edu), Patricia A. Keating (Dept. of Linguist, UCLA, Los Angeles, CA), Soo Jin Park (Dept. of Elec. Eng., UCLA, Los Angeles, CA), Shaghayegh Rastifar (Head and Neck Surgery, UCLA, Los Angeles, CA), and Abeer Alwan (Dept. of Elec. Eng., UCLA, Los Angeles, CA)

Increasing evidence suggests that voices are best thought of as complex auditory patterns, and that listeners perceive and remember voices with reference to a "prototype" or "average" for that talker. Little is known about how, and how much, individual talkers vary their voice quality across situations that arise in every-day speaking, so the nature and extent of variability underlying these abstract averages, and thus the nature of the averages themselves, is unclear. The theoretical relationship between acoustic similarity and confusability in the context of a prototype model also remains unclear. In this preliminary study, 9 tokens of the vowel /a/ were recorded from 5 females on three dates. Measures of F0, spectral slope, HNR, and formant frequencies and their variability were gathered for all voice samples and acoustic distances between talkers were calculated under the assumption that all acoustic variables were equally important perceptually. Perceptual confusability was assessed in a same/different task, and predictions under the equal perceptual weight assumption were tested. Discussion will focus on how much variability is required before a voice sample no longer sounds like the originating talker, and on how the perceptual importance of each acoustical variable varies across talkers and acoustic contexts. [Work supported by NSF and NIH.]

Session 4pUW

Underwater Acoustics: Three-Dimensional Underwater Acoustics Models and Experiments II

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Invited Papers

1:25

4pUW1. Out-of-plane effects in three-dimensional oceans. Michael B. Porter (HLS Res., 3366 N. Torrey Pines Court, Ste. 310, La Jolla, CA 92037, mikeporter@hlsresearch.com)

The two-dimensional beam tracing algorithms in the BELLHOP acoustic model have been extended to three dimensions in BELLHOP3D. The new model includes virtually all of the capabilities of the 2D model, including 1) spatially varying bottom-types, 2) eigenray calculations, 3) TL calculations, and 4) time series calculations. In addition, the 3D model includes both the geometric and Gaussian beam tracing options. Finally, a simple option change allows it to switch between 2D and 3D calculations to assess the effects of horizontal refraction. This talk will discuss the algorithm, demonstrate its capabilities, and present results that illustrate when horizontal refraction is important.

1:45

4pUW2. Numerically exact three-dimensional propagation. Ahmad T. Abawi (HLS Res., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, abawi@hlsresearch.com)

Solving propagation in a three-dimensional environment is one of the most challenging problems in computational physics. However, for cases where the environment possesses some kind of symmetry be it rotational or translational, the propagation problem can be simplified considerably. It can be shown that in problems where the environment is translationally invariant, the 3D wave equation can be Fourier transformed along the direction of translational symmetry to reduce it to a 2D equation for each spectral component. The 3D solution can be obtained by solving the 2D wave equation for each spectral component and performing the inverse Fourier transform. In this paper, we use the above technique to compute propagation in an ideal and a penetrable wedge. For the ideal wedge, the pressure-release boundary condition is applied to both boundaries and for the penetrable wedge, the pressure-release boundary condition is applied to the horizontal surface and the continuity of pressure and normal velocity is imposed on the sloped interface. To obtain a numerically exact solution, we use the virtual source technique to solve the 2D problem for each spectral component.

2:05

4pUW3. Coupled mode analysis of three-dimensional propagation over a cosine shaped hill. Megan S. Ballard (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

Three-dimensional propagation over an infinitely long cosine shaped hill is examined using an approximate normal mode/parabolic equation hybrid model that includes mode coupling in the out-going direction. The slope of the hill is relatively shallow, but it is significant enough to produce both mode-coupling and horizontal refraction effects. The modeling approach is described and the solution is compared to results obtained with a finite element method to evaluate the accuracy of the solution in light of assumptions made in formulating the model. Then, the calculated transmission loss is interpreted in terms of a modal decomposition of the field, and the solution from the hybrid model is compared to adiabatic and Nx2D solutions to assess the relative importance of horizontal refraction and mode-coupling effects. An analysis using a horizontal ray trace is presented to explain differences in the modal interference pattern observed between the 3D and Nx2D solutions. The detailed discussion provides a thorough explanation of the observed 3D propagation effects and demonstrates the usefulness of the approximate normal mode/parabolic equation hybrid model as a tool to understand measured transmission loss in complex environments. [Work supported by ONR.]

Contributed Papers

2:25

4pUW4. Modeling three dimensional scattering from rough ocean boundaries using finite elements. Marcia J. Isakson and Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

Scattering from three-dimensional rough interfaces can be estimated using approximations to the Helmholtz-Kirchhoff integral such as perturbation theory or the Kirchhoff approximation. However, these solutions are constrained to the boundary. Physical processes occurring beneath the boundary such as scattering from layers or volume inclusions require additional approximations. Finite element models can benchmark these approximations since scattering the entire volume is calculated. In this study, a three-dimensional finite element model of acoustic scattering from rough interfaces is developed and compared with boundary element solutions. This model is an improvement on the existing two-dimensional and longitudinally invariant models previously presented. The full three-dimensional nature of this solution allows realistic representations of layering and volume inclusions, including shells, air bubbles, and targets. [Work supported by ONR, Ocean Acoustics.]

2:40

4pUW5. Three-dimensional underwater acoustic scattering from ocean boundaries: Proposed benchmark problems. Marcia J. Isakson, Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu), and Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

Acoustic scattering from ocean boundaries is a major component of both the signal for mapping sonars and the noise for imaging sonars. However, models of scattering are often reliant on boundary methods such as the evaluation of the Helmholtz-Kirchhoff integral and its approximations. These methods do not account for sub-surface scattering from layers or inclusions although additional approximations are often used. Often, these approximations are employed with only a cursory knowledge of their realm of validity with respect to such parameters as interface roughness conditions, sediment types and layering structure. We propose a series of benchmark problems to begin to establish the regions of validity for extant scattering models. The problems include 3D scattering from an air/water interface, scattering from a layered ocean sediment and scattering from a simple target buried beneath a rough ocean floor. By comparing the results from a variety of benchmark models, it may be possible to establish a better range of validity for the current suite of modeling tools. [Work supported by ONR, Ocean Acoustics.]

2:55

4pUW6. Three-dimensional underwater sound propagation: Proposed benchmark problems. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu) and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

A set of benchmark problems is suggested for testing the accuracy and efficiency of numerical solutions for three-dimensional (3D) underwater sound propagation. The physical causes of the 3D propagation conditions considered in this talk include out-of-plane reflections from medium boundaries and interfaces and horizontal refractions caused by changes of the index of refraction in the medium. Five different propagation scenarios are suggested: (1) sound propagation in a slope environment, (2) nonlinear internal wave waveguides, (3) submarine canyons, (4) seamounts, and (5) 3D waveguides with surface waves and/or seafloor sand waves. Both idealized and realistic environments will be considered. To allow theoretical closed-form solutions, the model environments will be simplified to create idealized problems, such as an impenetrable wedge/slope model and a semi-circular waveguide with impenetrable boundaries. For the problems which do not have closed-form solutions, different numerical solutions will be generated for comparisons. The numerical methods focused in this talk include 3D parabolic-equation, normal mode and finite element methods, and discussions on other numerical techniques, such as ray tracing, wavenumber integration, finite difference, and boundary element methods, will also be provided. [Work supported by the ONR.]

3:10–3:25 Panel Discussion

3:25–3:40 Break

Invited Paper

3:40

4pUW7. Links between acoustic field statistics, baroclinic wave geometry, and bathymetry computed with time-stepped three-dimensional acoustic simulations. Timothy F. Duda, Arthur E. Newhall, and Ying-Tsong Lin (Woods Hole Oceanographic Inst., WHOI AOPE Dept. MS 11, Woods Hole, MA 02543, tduda@whoi.edu)

Simulations of sound propagation in a canyon region and in an area populated with internal-wave packets have been performed. Time dependence was introduced into the three-dimensional water-column environments using dynamical ocean models. In the canyon simulation, time-dependence is limited to waves with hydrostatic pressure generated by tidal transports prescribed at the dynamical model boundaries. The wave-packet simulation includes both hydrostatic and nonhydrostatic waves. For the canyon, propagation of 300-Hz sound from shallow to deep water is simulated. For wave packets, 200-Hz sound is simulated. For each situation, statistics related to horizontal correlation are estimated at synthetic “arrays” inserted into the domain. Correlation properties are estimated from single field snapshots and from time-series analysis of the output, and the two are compared. Array performance evaluations from the two methods are also compared. Covariance matrix properties for the time-series output, including principal component behavior, are analyzed to examine the interference effects and fluctuation mechanisms that are responsible for the spatially patchy array performance results.

4:00

4pUW8. Three-dimensional parabolic-equation solutions with boundary-fitted grids. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

A higher-order operator splitting method incorporating multi-dimensional cross terms has recently been proposed to solve the three-dimensional (3D) parabolic wave equation consisting of a square-root Helmholtz operator. The advantages of this splitting method include providing a more accurate 3D parabolic-equation (PE) approximation, as well as supporting fast marching solvers, such as the Alternating Direction Implicit (ADI) Padé method. To apply this numerical solution scheme to a 3D underwater acoustic waveguide with surface waves in a boundary-fitted model grid, one can employ a one-dimensional (1D) non-uniform discretization formula derived from Galerkin's method using asymmetric basis functions [W.M. Sanders and M.D. Collins, J. Acoust. Soc. Am. **133**, 1953–1958 (2013)]. The use of this discretization formula is to approximate the two alternating sets of 1D differential equations with respect to either one of the two transverse directions at the ADI marching steps. An idealized semi-circular waveguide problem with a closed-form solution is considered as a benchmark to test different 3D PE solutions. Compared to the fixed grid PE solution, the boundary-fitted PE solution is an order of magnitude more accurate in terms of both the magnitude error per unit distance and the relative phase error. [Work supported by the ONR.]

4:15

4pUW9. Three-dimensional noise modeling in a submarine canyon. David R. Barclay (Dept. of Oceanogr., Dalhousie Univ., PO Box 15000, Halifax, Nova Scotia B3H 4R2, Canada, dbarclay@dal.ca) and Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The ambient sound field due to wind generate surface noise in an idealized Gaussian submarine canyon can be described using the method of normal mode decomposition applied to a three-dimensional longitudinally invariant wave-guide. The modal decomposition is carried out in the vertical and across-canyon horizontal directions and gives a semi-analytical solution describing the three-dimensional topographic effects on the spatial distribution of the noise, the power spectrum, and the vertical and horizontal coherences. Additionally, the noise field can be computed using wave-equation reciprocity and either a three-dimensional cylindrical co-ordinates parabolic equation (PE) or an Nx2D PE sound propagation model. Inter-comparison of these models highlights the effect of the three-dimensional topography on the vertical coherence and mean-noise level as a function of arrival direction relative to the canyon's axis. These effects include the focusing of noise along the canyon axis and the frequency perturbation of vertical coherence minima.

4:30

4pUW10. Time-domain broadband simulations of arbitrarily shaped three-dimensional elastic targets on water/sand interface. Yu Shao, Myoung An, and Shumin Wang (Auburn Univ., 200 Broun Hall, Auburn, AL 36849, wangs@auburn.edu)

Acoustic wave scattering from buried or proud elastic targets on ocean floor is important to a number of engineering problems. Existing simulation approaches are mainly based on frequency-domain finite-element method (FEM). Although accurate, it is inefficient for broadband acoustic signals for two main reasons. First, the computational cost is proportional to the number of frequency points of interests. Second, inversion of the resulting FEM matrix at each frequency point can be quite time-consuming as the number of unknown increases. The latter can be especially problematic at higher frequencies as the spatial resolution or element order increases. A staggered-grid finite-difference time-domain (FDTD) method is developed to overcome these problems. It simulates the time-domain signals directly in a leap-frog fashion without the needs of explicitly storing and inverting system matrices. The perfectly matched layers (PML) technique is applied to truncate the simulation domain and the total-field/scattered-field approach is employed to incorporate incident, scattered, and transmitted waves from the water/sand interface. Finally, the far-field scattered field is computed via the layered Green's function from the near-field results. Several three-dimensional numerical examples are further provided to demonstrate its accuracy and efficiency.

4:45

4pUW11. Parameter dependence of normal modes for a coastal front model. Brendan J. DeCourcy (Mathematical Sci., Rensselaer Polytechnic Inst., 1521 6th Ave., Apartment 313, Troy, NY 12180, decoub@rpi.edu), Ying-Tsong Lin, James F. Lynch (Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA), and William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY)

A feature model for a shallow ocean front over a bottom with constant slope [Y.-T. Lin and J. F. Lynch, J. Acoust. Soc. Am. **131**, EL1–EL7 (2012)] is analyzed to determine the parameter dependence of three-dimensional normal mode solutions. The front is a curved interface between two isospeed regions in a coastal wedge. Relevant parameters are the front distance from the straight shoreline, bottom slope angle, sound speeds, and source frequency. The cross-slope wavenumbers are determined by a complicated dispersion relation involving Hankel functions, so a series of asymptotic approximations is applied to simplify the equation. The result is converted into a first-order differential equation for the wavenumber variation with respect to a selected parameter, and accurate and explicit solution expressions are obtained. These expressions permit convenient specification of how model parameters influence acoustic quantities such as modal phase speed. This approach is designed for application to other ocean feature models which contain large or small dimensionless combinations of parameters. The objectives are to help explain how acoustic metrics depend on environmental and geometrical parameters, and to assist in assessing results from integrated ocean-acoustics models. [Work supported by the ONR.]

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees of the Acoustical Society of America will hold open meetings on Tuesday and Thursday evenings. On Tuesday, the meetings will begin at 7:30 p.m., except for Engineering Acoustics, which will hold its meeting starting at 4:30 p.m. On Thursday evening, the meetings will begin at 8:00 p.m. or 8:30 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 Thursday are as follows:

8:00 p.m.	
Biomedical Acoustics	Kings 2
Musical Acoustics	Brigade
Noise	King 5
Speech Communication	Commonwealth 2
Underwater Acoustics	Ballroom 4
8:30 p.m.	
Signal Processing in Acoustics	Ballroom 3

Session 5aBA

Biomedical Acoustics: Medical Ultrasound

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

Contributed Papers

8:00

5aBA1. Numerical evaluation of absorbing boundary layers for the transient Khokhlov–Zabolotskaya–Kuznetsov Equation. Xiaofeng Zhao and Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI 48824, zhaoxia6@msu.edu)

FOCUS, the "Fast Object-Oriented C++ Ultrasound Simulator" (http://www.egr.msu.edu/~fultras-web), simulates nonlinear ultrasound propagation in the time-domain by numerically evaluating the transient Khokhlov–Zabolotskaya–Kuznetsov (KZK) equation. KZK simulations in FOCUS previously required a computational grid with a large radial distance relative to the aperture radius to reduce the effect of reflections from the boundary. To decrease the size of the grid required for these calculations, an absorbing boundary layer derived from the power law wave equation provides an alternative to the stretched coordinate system in a perfectly matched layer. Simulations of the linear pressure fields generated by a spherically focused transducer with an aperture radius of 1.5 cm and a radius of curvature of 6 cm are evaluated for a short pulse with a center frequency of 1 MHz and a peak surface pressure of 0.5 MPa. Numerical results for linear KZK simulations with and without the absorbing boundary layer are compared to an on-axis analytical solution. Results of nonlinear KZK simulations with and without the PML are also evaluated, and all of these show that the absorbing layer effectively attenuates the wavefronts that reach the boundary of the computational grid. [This work was supported in part by NIH Grant R01 EB012079.]

8:15

5aBA2. An improved interpolation approach for rapid simulations of pulse-echo ultrasound imaging. Leslie P. Thomas and Robert McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI 48824, lt.thomas.jr@gmail.com)

New B-mode image simulation routines in FOCUS (http://www.egr.msu.edu/~fultras-web) apply linear interpolation to a finely sampled pressure signal that is calculated once per scatterer and then reused for different time delay and amplitude values in subsequent A-lines. This approach achieves more accurate results in less time than simulations that calculate the pressure waveforms on a coarser grid followed by cubic spline interpolation. To demonstrate this result, simulations are performed using 24 element sub-apertures in a linear array consisting of 192 rectangular elements that are 5 mm high and 0.5133 mm wide with 0.1 mm center-to-center spacing. The center frequency of the excitation is 3 MHz for simulations of a computer phantom with 100,000 scatterers. Results for each approach are compared to a simulated reference signal calculated with the impulse response for a sampling frequency of 1 GHz. To achieve an RMS error of 6%, FOCUS requires linearly interpolated signals sampled at 16 MHz, where the stored signal was sampled at 64 MHz. The total computation time for the linear interpolation approach was 36 min, whereas the cubic spline interpolations required a frequency of 20 MHz and took 49 minto run to achieve a similar RMS error. [This work was supported in part by NIH Grant R01 EB012079.]

8:30

5aBA3. Toward monodisperse ultrasound-triggered phase-shift emulsions using differential centrifugation. Kyle Stewart (Biomedical Eng., Univ. of Cincinnati, 231 Albert Sabin Way, Cardiovascular Ctr., Rm. 3972, Cincinnati, OH 45267-0586, stewake@mail.uc.edu), Kirthi Radhakrishnan, and Kevin J. Haworth (Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Acoustic droplet vaporization (ADV) is a process that enables the *in situ* production of microbubbles from an injected perfluorocarbon emulsion and it has been investigated for imaging and therapeutic applications. High-speed mechanical shaking rapidly produces a polydisperse emulsion (~10^{10} droplets/mL) with droplets ranging from less than 400 nm to greater than 15 μm. The ADV pressure amplitude threshold is higher and fraction of transitioned droplets lower for droplets smaller than approximately 2 μm in diameter. Droplets greater than approximately 8 μm in diameter are not suitable for systemic administration. Therefore, high-speed mechanical shaking produces many droplets of limited utility. Differential centrifugation has been used as a size isolation technique for polydisperse ultrasound contrast agents. By applying a similar technique to a perfluorocarbon emulsion, the volume-weighted fraction of droplets between 2 and 5 μm was increased from 29$\pm$2% to 93$\pm$3%. The transition efficiency of the droplets between 2 and 5 μm was the same regardless of whether a polydisperse or monodisperse distribution was insonfied with 2 MHz ultrasound. The ADV pressure threshold of differentially centrifuged droplets was similar to non-centrifuged droplets. [Supported in part by NIH grant KL2 TR000078.]

8:45

5aBA4. Characterizing the pressure field in a modified microbubble flow cytometer: Using a laser Doppler vibrometer to validate the internal pressure. Cheng-Hui Wang (Inst. of Appl. Acoust., Shaanxi Normal Univ., Xi'an, Shaanxi, China), Camilo Perez (BioEng. and Ctr. for Industrial and Medical Ultrasound - Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Appl. Phys. Lab. - CIMU, Seattle, WA 98105-6698, camipiri@uw.edu), Jarred Swalwell (Oceanogr., Univ. of Washington, Seattle, WA), Brian MacConaghy (Ctr. for Industrial and Medical Ultrasound - Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Juan Tu (Key Lab. of Modern Acoust., Nanjing Univ., Nanjing, China), and Thomas J. Matula (Oceanogr., Univ. of Washington, Seattle, WA)

Previously, a flow cytometer was modified with a PZT transducer in order to study the radial oscillations of statistically significant numbers of microbubbles (J. Acoust. Soc. Am. **126**, 2954–2962, (2009)). We reported the results of pressure calibration for transient sonication in a recent symposium (ASA, Indianapolis, 2013). Here, we report the results of pressure calibration for steady-state sonication. Because the flow channel width (< 200 μm) is too narrow to insert our hydrophones, we rely on finite element analysis (FEA) to predict the acoustic pressure field. In this study the simulation results were compared to Laser Doppler Vibrometer *in-situ* measurements of the velocities of the surface of the flow chamber. The OFV-534 sensor head with OFV-5000 controller (Polytec, Irvine CA) was mounted so that the laser reflected off the proximal outer surface of the flow chamber. The FEA model coupled structural vibration and linear acoustic physics to calculate the steady state pressure. The FEA model compared favorably with the LDV measurements. The FEA simulations were used to predict the pressure field, leaving only the shell elasticity and viscosity χ and κ as unknown variables in the bubble dynamics model. Excellent fits to Optison bubble oscillations were obtained.

9:00

5aBA5. The features of sound propagation through human lungs, revealed by transmission sounding with phase manipulated acoustic signal of 80–1000 Hz frequency band. Vladimir Korenbaum and Anton Shiryaev (Pacific Oceanologic Inst., 43, Baltiiskaya Str., Vladivostok 690041, Russian Federation, v-kor@poi.dvo.ru)

The sound propagation in human lungs remains poor studied. We studied the phenomenon in a frequency range of 80–1000 Hz by means of sounding lungs with phase manipulated signal injected into mouth and right/left supraclavicular chest areas. The installation included the subwoofer (mouth) and the shaker (supraclavicular zone), both fed through the power amplifier from the sound card of laptop, and a system of accelerometer sensors, connected to PowerLab (ADInstruments). Transmitted signals were recorded above medial trachea and in eight medial/basal posterior chest positions of 11 volunteers. Distances emitter-sensor and sensor-sensor were measured by pelvimeter. Convolution technique was applied for signal processing. Two to three arrivals of the signal were measured above trachea for sounding through mouth. The second arrival is treated as a reflection through air lumen from distal bronchi level with the distance estimated as 23 ± 5 cm (sound speed 200 m/s). Four to five arrivals with various median velocities were measured in posterior chest positions for sounding through mouth as well as both supraclavicular chest areas. The air-structural (air—tissue) and pure structural (tissue) transmission paths are identified for sounding through mouth. Meanwhile only pure structural transmission paths are supposed for chest sounding. [This study was supported by RFBR grant 13-08-00010.]

9:15

5aBA6. Simultaneous measurement of sound pressure and temperature of tissue mimicking material by an optical fiber Bragg grating sensor. Keisuke Imade, Daisuke Koyama, and Iwaki Akiyama (Doshisha Univ., 1-3 Tataramiyakodani, Kyotanabe 610-0321, Japan, duo0315@mail4.doshisha.ac.jp)

An optical fiber Bragg grating (FBG) sensor is capable of simultaneous measurement of sound pressure and temperature. The FBG is a type of refractive index grating constructed in a short segment of optical single mode fiber that reflects particular wavelengths of light. This Bragg wavelength depends on both of the sound pressure and the temperature. We reported the simultaneous measurement of sound pressure and temperature in water under the conditions of medical application. In this study, temperature rise of a tissue mimicking material (TMM) caused by exposure to ultrasound in the range of 0.3 to 5 MPa were measured. The TMM is made of agar-gel with Glycerin solution of 11.21%. The FBG of 1 mm in length constructed in the optical fiber of 0.25 mm in diameter was used in the experiments. As a result, as the measured sound pressure was 5 MPa, the measured temperature rise was 6.0 degree. Thus this sensor is capable of simultaneous measurement of sound pressure and temperature rise of biological tissues exposed to the ultrasound. [This study was supported by MEXT-Supported Program for the Strategic Research Foundation at Private Universities, 2013–2017.]

9:30

5aBA7. On the use of local speckle field as a correction factor for shear modulus estimates based on multiple-track-locations methods. Laurentius O. Osapoetra (Phys. & Astronomy, Univ. of Rochester, 500 Joseph C Wilson Blvd. CPU Box 271443, Rochester, NY 14627, loscar@pas.rochester.edu), Jonathan H. Langdon, and Stephen A. McAleavey (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Acoustic-radiation-force impulse (ARFI) imaging for characterization of shear modulus of biological tissues employs either multiple-track-locations (MTL) methods or single-track-location (STL) methods. MTL estimates of shear modulus at different depths suffer from more variability compared to those of STL estimates. Our studies have shown a significant correlation between bias in shear-wave arrival time estimates and local speckle field lateral statistics. We propose using the local speckle field as a surrogate for the unknown bias apparent in MTL shear-wave speed estimates. This local speckle field is determined using a "swept-receive acquisition" that is produced by holding the transmit beam fixed while laterally translating the receive aperture. Application of various lateral weighting functions to the swept-receive image results in an approximate compensation to the tracking bias. In this study, we implement our technique on simulation and experimental data from gelatin phantoms. Additionally, the effect of varying transmit-receive aperture f/# on the accuracy of our compensation method is investigated in simulation. Finally, the quality of compensated MTL shear modulus images are quantified and compared to those of the uncompensated MTL and of STL methods in matched tissue mimicking phantom experiments.

9:45

5aBA8. Iterative reconstruction of the ultrasound attenuation coefficient from backscattered signals. Natalia Ilyina (Dept. of Cardiovascular Sci., KU Leuven, UZ Herestraat 49 - box 7003 50, Leuven, België 3000, Belgium, natalia.ilyina@uzleuven.be), Jeroen Hermans (DoseVue, Hasselt, Belgium), Erik Verboven, Koen Van Den Abeele (Dept. of Phys., KU Leuven Kulak, Kortrijk, Belgium), Emiliano D'Agostino (DoseVue, Hasselt, Belgium), and Jan D'hooge (Dept. of Cardiovascular Sci., KU Leuven, Leuven, Belgium)

Estimation of the local acoustic attenuation from backscattered signals has several clinical applications but remains an open problem. Most of the proposed solutions relate the observed spectral changes directly to the theoretical predictions. However, these methods make a number of approximations and require correction strategies for acoustic wave phenomena that are not accounted for in the model (e.g., non-linearity). In this study, attenuation was estimated by successively solving the forward wave propagation problem for different attenuation coefficients and by matching the calculated backscattered signals to the observed one. For the forward problem, the effects of attenuation, nonlinear distortion, reflection and scattering were taken into account. The proposed approach was validated on simulated data and data recorded in six tissue mimicking phantoms and was compared to conventional methods. The relative error of the attenuation coefficient remained below 10% for the simulated and phantom data. The conventional methods showed a comparable performance on the simulated data, but their error significantly increased in the phantom study. The proposed method outperformed state-of-the-art attenuation estimators. Moreover, it can be used to estimate local non-linearity. In future work, the propagation model will be extended to 3D and diffraction effects will be included.

10:00–10:15 Break

10:15

5aBA9. Non-invasive monitoring of Achille's tendon stiffness variations *in-vivo* using mechanical vibrations. Muhammad Salman (Systems and Mech. Eng., Southern Polytechnic State Univ., 1100 S Marietta Pkwy # G-172, Marietta, GA 30060, msalman@spsu.edu) and Karim Sabra (Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

A non-invasive monitoring technique of laser Doppler vibrometer (LDV) is used to find the stiffness of Achilles tendon. It is difficult for an ultrasound to collect the elastographic images of high stiffness areas such as Achilles tendon. Magnetic resonance images (MRI) technique is expensive and requires extensive training of the clinicians for elastographic image processing. This non-invasive technique needs short setup time and a simple physical structure for the data collection. A shaker is used as an excitation source, which generates waves on the tendon surface. This dynamic elastography technique measure wave velocities by an LDV. Achilles tendon is excited from 10 Hz to 1000 Hz using shaker and sensed by the LDV at four positions, which are one cm apart. Cross correlation signal processing is used for finding the time delays of the waves approaching each sensor location. It is found that as the contraction level increases, tendon stiffness increases. A comparison of average and varied stiffness values is shown in Achilles tendon. This technique may assist clinicians in characterizing muscle tone changes due to sport injuries in tendon.

10:30

5aBA10. Empirical mode decomposition-discrete Hilbert transform based solution to detect decompression-induced gas bubble from Doppler ultrasound signal. Md Iqbal Aziz Khan and Takayoshi Nakai (Graduate School of Sci. and Technol., Shizuoka Univ., Hamamatsu, 3-5-1 Johoku, Naka-ku 432-8561, Japan, iqbal_aziz_khan@yahoo.com)

This paper concerns the detection of decompression-induced gas bubble from underwater construction workers Doppler ultrasound signal based on empirical mode decomposition (EMD), discrete Hilbert transform (DHT), and some parameters, which could be useful to detect gas bubble. EMD in combination with DHT is used to generate time-frequency-energy distribution and systolic phase is detected from that distribution. The properly detection of the systolic phase is the most important task to the detection of gas bubble. Time-varying mean frequency (TMF) is determined from instantaneous amplitude and instantaneous frequency. An algorithm is applied to extract some specific segments from TMF. Then the segments are considered as gas bubble on the basis of some parameters such as spreading in frequency, ratio of the gas bubble signal to the background signal, ratio of the gas bubble signal's total energy to the background signal, and rising rate.

10:45

5aBA11. Three-dimensional pulsation of rat carotid artery bifurcation observed using a high-resolution ultrasound imaging system. Changzhu Jin, Kweon-Ho Nam, and Dong-Guk Paeng (Ocean System Eng., Jeju National Univ., Jeju, Jeju Self Government Province 690-756, South Korea, yustchang@gmail.com)

The arterial structure experiences cyclic pulsation in three-dimension (3-D) by pulsatile blood flow. Here we reconstruct the 3-D carotid artery bifurcation (CAB) geometry from three rats to observe the pulsatile variation of the carotid artery geometry by a high-resolution ultrasound imaging system (Vevo 770, VisualSonics, Canada). Two experienced observers manually segmented the cross-sectional ultrasound images to outline the arterial lumen. From the constructed geometry of three subjects, we observed that the CAB geometry favorably expanded to anterior/ posterior direction which parallel to sagittal plane. Furthermore, an *in vitro* blood flow experiment using an anthropomorphic flow phantom was implemented to more clearly understand the asymmetric pulsation phenomenon. Finally we confirmed the elastic bifurcation geometry favorably expands in a direction of bifurcation, both *in vivo* and *in vitro* measurements, which derived by bifurcated flow. This finding, the asymmetrical pulsation phenomenon of carotid bifurcation in 3-D, may be useful in understanding hemodynamic etiology of cardiovascular diseases and also provide more realistic input data for computer simulation of hemodynamics. [This work was supported by MSIP Korea, under the C-ITRC support program (NIPA-2014-H0401-14-1002) supervised by the NIPA and also supported by the Richard Merkin Visiting Fellowship Program of the Focused Ultrasound Foundation.]

11:00

5aBA12. Ability of skeletal muscle to protect bones and joints from external impacts: Acoustical assessment. Armen Sarvazyan (Artann Labs., 1459 Lower Ferry Rd., Trenton, NJ 08618, armen@artannlabs.com), Sergey Tsyuryupa (Artann Labs., Lambertville, NJ), and Oleg Rudenko (Phys. Dept., Moscow State Univ., Moscow, Russian Federation)

One of the major (but the least studied!) functions of skeletal muscle is protecting the skeletal system from external impacts by absorbing and redistributing the energy of mechanical shock in time and space. During muscle contraction, its elasticity modulus is greatly increased, which partly unloads adjacent bones and skeletal joints. Muscle viscosity is also greatly increased helping to absorb and dissipate dangerous shocks. Elasticity and viscosity data may be obtained using the measurement of shear wave velocity and attenuation. In this study, we investigated changes in the velocity and attenuation of shear acoustic waves in an anisotropic tissue phantom mimicking skeletal muscle under different level of tension of the fibers imbedded in the phantom. Stretching the fibers simulates the muscle contraction. It is shown that both velocity and attenuation of shear waves propagating along the fibers significantly increase with fiber tension while they are negligibly affected in the case of wave propagation across the fibers. We developed a theory for propagation of shear waves in anisotropic medium simulating the muscle and muscle contraction. Equations for the speed and attenuation of various modes of shear acoustic waves are derived. Theoretical predictions are in agreement with experimental data. [NIH R21AR065024.]

11:15

5aBA13. Novel use of ultrasound imaging to decode activity of forearm muscles for upper extremity prosthetic control. Nima Akhlaghi (Elec. and Comput. Eng., George Mason Univ., 4400 University Dr., Fairfax, VA 22030, nakhlagh@gmu.edu), Mohamed Lahlou (BioEng., George Mason Univ., Fairfax, VA), Brian J. Monroe (Hanger Clinic, Fairfax, Virginia), Parag V. Chitnis (BioEng., George Mason Univ., Fairfax, VA), Huzefa Rangwala, Jana Kosecka (Comput. Sci., George Mason Univ., Fairfax, VA), Joseph J. Pancrazio, and Siddhartha Sikdar (BioEng., George Mason Univ., Fairfax, VA)

With the recent developments in the electro-mechanical design of upper extremity prosthetics, the need for more advanced control strategies for such prosthetics has increased. Current commercially available prostheses based on myoelectric control have limited functionality, which leads to many amputees abandoning use. Myoelectric control using surface electrodes has a number of limitations, such as low signal to noise ratio and lack of specificity for deep muscles. To address these limitations, and enable more intuitive dexterous control, we propose a new strategy for sensing muscle activity based on real-time ultrasound imaging. Ultrasound imaging of the forearm muscles was performed on six healthy volunteers and a transradial amputee using a Sonix RP system with a 5–14 MHz linear array transducer. Images were analyzed to map muscle activity based on the changes in the ultrasound echogenicity of the contracting muscles during different complex movements, and used to control a virtual prosthetic hand. Individual digit movement could be decoded with 97% accuracy, and 15 different complex grasps could be decoded with 87% accuracy. In a transradial amputee, we were able to differentiate between seven different movements. These preliminary results demonstrate the feasibility of using ultrasound imaging for control of upper extremity prostheses.

11:30

5aBA14. Validity of acoustic method for the assessment of whole-body hydration status. Alan Utter, Mason C. Calhoun, Steven R. McAnulty, Jeffrey M. McBride, Jennifer Zwetsloot, Melanie Austin, Jonathan D. Mehlhorn, Lesley Sommerfield (Health and Exercise Sci., Appalachian State Univ., 111 Rivers St., Boone, NC 28608, utterac@appstate.edu), Sergey Tsyuryupa (Artann Labs., Trenton, NJ), and Armen Sarvazyan (Artann Labs., Lambertville, NJ)

In almost any sport, athletes undergoing dehydration often suffer from numerous dehydration related injuries. The purpose of this study was to evaluate the validity of acoustic method to detect changes in the hydration status of athletes after undergoing acute dehydration and a 2-hour rehydration protocol. The acoustic method of assessing body hydration status is based on the experimental fact that ultrasound speed in muscle is a linear function of the tissue water content. The assessment of water imbalance was conducted by measuring speed of ultrasound in the calf muscles using through transmission method. Eighty-two male and female collegiate athletes were examined to detect changes in hydration status before and after undergoing 3% acute dehydration. Results demonstrated that the changes of ultrasound velocity are in average about 1.1 m/s per 1% of body dehydration and ultrasound velocity in muscle potentially may serve as a measure of body hydration status. However, ultrasound speed measurement using through transmission mode implemented in this study is highly dependent on the positioning of the probe: even slight variation in the acoustic path results in significant changes in the measured values, which may results in unacceptable error. A solution to this problem is proposed and discussed. [NIH2R44AG042990.]

11:45

5aBA15. Design and characterization of a sensitive optical micro-machined ultrasound transducer. Suzanne M. Leinders (Acoust. Wavefield Imaging, Delft Univ. of Technol., Lorentzweg 1, Delft 2628 CJ, Netherlands, s.m.leinders@tudelft.nl), Wouter J. Westerveld (Optics Res. Group, Delft Univ. of Technol., Delft, Netherlands), Jose Pozo, Paul van Neer (TNO, Tech. Sci., Delft, Netherlands), H. P. Urbach (Optics Res. Group, Delft Univ. of Technol., Delft, Netherlands), Nico de Jong, and Martin D. Verweij (Acoust. Wavefield Imaging, Delft Univ. of Technol., Delft, Netherlands)

Novel 3D intravascular or transesophageal ultrasound approaches require transducer arrays containing many small elements. Conventional piezo-electric techniques face fabrication challenges due to narrow kerfs and dense wiring. Micro-machined alternatives like CMUTs and PMUTs lack either sensitivity or bandwidth to fully compete. Therefore we developed an opto-mechanical ultrasound sensor. The absence of wiring makes it MRI compatible. The developed OMUT contains integrated photonics, which is fabricated using standard silicon-on-insulator technology, providing a small footprint and enabling mass production and ease of integration. The sensor consists of a straight waveguide and a micro-ring resonator integrated on a 124 μm wide, 2.7 μm thick acoustical membrane. Light, passing the waveguide, is partly coupled into the ring resonator. A dip appears in the spectrum of the transmitted light at the resonance wavelength of the micro-ring. If the acoustical membrane and hence the micro-ring is deformed due to an incident ultrasound wave, this is observed as a shift in the resonance wavelength of the ring. This paper presents the construction and characterization of our device. Measurement results of the linearity, frequency response, sensitivity and temperature dependence are compared with a model. The results demonstrate that our OMUT is a promising basis for future ultrasound arrays.

FRIDAY MORNING, 22 MAY 2015 KINGS 4, 8:30 A.M. TO 11:50 A.M.

Session 5aMU

Musical Acoustics: Non-Western Musical Instruments and Performance Spaces

Jonas Braasch, Chair
School of Architecture, Rensselaer Polytechnic Institute, 110 8th Street, Troy, NY 12180

Chair's Introduction—8:30

Invited Papers

8:35

5aMU1. Understanding timbral effects of multi-resonator/generator systems of wind instruments in the context of western and non-western music. Jonas Braasch (Ctr. for Cognition, Commun., and Culture, Rensselaer Polytechnic Inst., 110 8th St., Troy, NY 12180, braasj@rpi.edu)

Wind instruments are often modeled as a coupled generator-resonator system, which in case of the clarinet consists of a reed and a cylindrical pipe. However, the fact that the throat and oral cavities behind the tone generator also play a critical role is often neglected. While the resonance pipe of a wind instrument can be adjusted in length through keyholes or valves, its width is fixed. The opposite can be observed for the human resonance system where the length is fixed but the dimensions of the cross-section are variable. This is important fact is routinely used to shape the timbral qualities of tones and to determine the tones' fundamental frequencies—for example, to generate high notes above the normal range on a saxophone. In many music cultures, time-varying timbral modifications are more important than melodic aspects, for example, in traditional digeridoo practice. A third resonator system is the room the musical instrument is performed in. This can also have important timbre shaping aspects, emphasizing, for example, the brilliant sound of Bach trumpets. An extended saxophone [Braasch, 2014, J. Acoust. Soc. Am. **135**, 2245] and convolution reverb is used to demonstrate and analyze the effects of multi-resonator/generator systems for this presentation.

8:55

5aMU2. "Good acoustics" is culturally determined: Evidence that prehistoric performance space selection was based on different world views. Steven J. Waller (Rock Art Acoust., 5415 Lake Murray Blvd. #8, La Mesa, CA 91942, wallersj@yahoo.com)

"Ideal" performance spaces are in the ear of the listener. Cases will be presented to argue that the response to a particular set of acoustic characteristics of a space is a function of the world view of the audience/performers. For example, in today's modern society that values speech intelligibility and appreciation of polyphonic classical and contemporary music, a distinct echo in a performance space is considered an unforgivable fault. However, in the ancient past when wave characteristics of sound were not understood, virtual sound effects arising from complex sound wave interactions (echoes, reverberations, interference patterns, etc.) were misinterpreted as invisible beings (echo spirits, thunder gods, sound absorbing bodies, etc.) as described in ancient myths around the world. Some of these myths contain accounts of purposeful searches for the best echoes, which were worshiped. Scientifically conducted experiments involving blindfolded participants show how various ambiguous sounds can be interpreted in more than one way (like optical illusions). Sound level measurements demonstrate that cave paintings and canyon petroglyphs were placed in locations with the strongest echoes. Prehistoric flutes have been found in reverberating caves. These experiments thus can help in understanding our ancestors' perceptions and reactions to sounds they considered supernatural (https://sites.google.com/site/rockartacoustics/).

9:15

5aMU3. The origins of building acoustics for theater and music performances. John Mourjopoulos (Audio & Acoust. Technol. Group, Elec. and Comput. Eng., Univ. of Patras, Audio Group, WCL, Electricak and Comput. Eng. Dept. University of Patraser E, Patras 26500, Greece, mourjop@upatras.gr)

The ancient Greek amphitheaters and roofed odeia represent the earliest examples of acoustics utilized for enhancing theatrical and music performances over large public audiences, often up to 15,000 participants. Such an early achievement, more than 2000 years ago, was possibly crucial for the foundation of these performance-based art forms within the ancient Greek society and the eventual geographical spread of these buildings. Through the continuous evolution during the Roman era, via the Renaissance theaters and the modern concert halls, these specially constructed spaces allowed the performance-based art forms to be sustained and evolve as essential constituents of the western cultural heritage and civilization. We shall first consider some historical and architectural properties of these early performance spaces and then summarize their design principles and acoustic parameters. Acoustic properties were different for spaces used for theatrical and music performances: open-air theaters for drama were manly utilizing early reflections (e.g., up to 40 ms) to achieve perfect speech intelligibility for listener distances even beyond 60 m; in contrast, roofed-odeia had acoustics appropriate for music with prominent reverberation comparable to that of modern concert halls. The presentation will include auralization demos for important ancient theaters and odeia.

9:35

5aMU4. Conch-shells, bells, and gongs in Hindu temples. Marehalli Prasad (Mech. Eng., Stevens Inst. of Technol., 519 Hudson St., Carnegie Bldg., Hoboken, NJ 07030, mprasad@stevens.edu)

Acoustics plays an important role in Hinduism and Hindu temples and worship practices. Several instruments such as conch-shells, bells, and gongs due to their musical qualities are used to enhance the spiritual experience of devotees in Hindu temples. These instruments are used to augment the acoustically rich Vedic chanting and singing. These instruments are generally used in a room called *Ardha Mantapa* (in Sanskrit), which is coupled to the Sanctum Sanctorum (called as *Garbha Griha* in Sanskrit) in a Hindu temple. The *Garbha Griha* and the *Ardha Mantapa* play an important role in the acoustical environment in the temples. These instruments are used both individually as well as collectively during the worship. These instruments through their tonal quality complement the soundscape of the environment. This presentation deals with spectral measurements of these instruments and also the acoustical aspects of the temple space. Measurements taken in two Hindu temples will be presented.

9:55

5aMU5. On the acoustics of Maya pyramids. David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683-4514, dlubman@dlacoustics.com)

On my first visit to the ancient Maya ceremonial city of Chichen Itza in 1998, I discovered an amazing sonic phenomenon at a limestone pyramid known as the temple of Kukulkan. This four-sided 30 m high pyramid resides in a large open plaza with no nearby structures to produce echoes. But clap your hands at the pyramid and you hear a brief chirped echo. This was unexpected. Normally, echoes are delayed replica of their stimuli (a single handclap in this case). Yet this echo sounds unlike the handclap. It is a downward-gliding and harmonic-rich chirp. I recorded the echo for later study. Surprises continued when sonograms of the chirped echo were compared with sonograms of the quetzal—a magnificent bird venerated by the Maya and other Mesoamericans. The match was near-perfect! Was this a bizarre coincidence or a stunning archaeoacoustic discovery overlooked by archaeologists? The mathematics of the physical phenomena and the psychophysics of the chirped echo are now understood. Cultural evidence for intentional acoustical design by ancient Mesoamericans is strongly indicated. It seems certain that chirped echoes were the rule at staircased pyramids throughout Mesoamerica. Chirps weaken and become inaudible as limestone staircases erode and become porous.

10:15–10:30 Break

10:30

5aMU6. Traditional and recent performance practice in Asian free reed mouth organs: The sheng and khaen as case studies. James P. Cottingham (Phys., Coe College, 1220 First Ave., Cedar Rapids, IA 52402, jcotting@coe.edu)

Mouth-blown instruments employing a free reed coupled to a pipe resonator have long been used throughout East and Southeast Asia. Details of the origin and development of these instruments are not known, but are closely connected with the history and prehistory of a multitude of ethnic groups. Beginning from presumed folk instrument origins, the free reed mouth organs have been used in a variety of contexts including simple signaling, courtship, local entertainment, civic or military processions, and sophisticated court music. Two instruments operating on similar acoustical principles have had contrasting histories: the Chinese sheng and the Laotian khaen. The sheng has a two thousand year recorded history in China, and in the last century modernized versions have been developed and appeared in the Western concert hall style setting of the Chinese orchestra. The khaen, while remaining a strong cultural symbol of the Lao people, has not undergone similar developments as once prevalent traditional performance styles have almost disappeared.

10:50

5aMU7. Outdoor oratory and performance space. Braxton B. Boren (Mech. and Aerosp. Eng., Princeton Univ., 30-91 Crescent St., 5B, Astoria, New York 11102, bbb259@nyu.edu)

Spoken sermons and oratory in the West have usually been confined to the interior of churches or parliaments that bear similarities to traditional concert halls. However, the largest crowds in history are described as being outside due to the size constraints of most interior spaces. Historical estimates of crowd sizes or the intelligible range of speakers have generally been quite speculative. However, recent work has used an acoustical experiment by Benjamin Franklin to estimate the 1 m on-axis average SPL of the Anglican preacher

George Whitefield at 90 dBA. Computational simulation of Whitefield's preaching in London confirms Franklin's estimate that he could have been heard by crowds of 30,000 or more depending on weather conditions and crowd noise. Using Whitefield as a benchmark, other historical orations may be evaluated based on geometric and acoustic factors. Within this framework, speeches given by Pericles, Demosthenes, Alexander the Great, and Julius Caesar are examined based on the size and noise of the crowd as well as the level attainable by each speaker. In each case, outdoor oratory possesses unique performance considerations distinct from interior or musical settings.

11:10

5aMU8. The Paul's Cross sermons—Out of the cathedral and into the public square. Matthew Azevedo (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, mazevedo@acentech.com)

Paul's Cross, located in the northeastern yard of St. Paul's Cathedral, was the site of public sermons which were integral to the social fabric of early modern London. These sermons brought together a wide swath of London's population, "a great Congregation of thy Children... of all sorts... from the Lieutenant of thy Lieutenant, to the meanest sonne of thy sonne" in John Donne's words, for an event which served as a conduit from the King, through the Church, and out to the people of England. Early modern sermons were not merely lectures, but were theatrical conversations between the preacher and his congregation in which each had an active role to play, staged for their entertainment value as well as to elevate the spirit and the mind. The Paul's Cross Project attempts to understand this interplay between preacher and congregation in an open, public space through acoustical and visual modeling of St. Paul's churchyard as it existed before the Great Fire of 1666. The acoustical modeling has resulted in an auralization of the sermon which allows listeners to directly experience what it might have been like to attend one of the Paul's Cross sermons.

11:30

5aMU9. Recording studios—Optimized, contrived, and augmented spaces. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

Multitrack production occurs around the globe in spaces from large to small. Their acoustic capability ranges from thoughtfully designed purpose-built spaces for recording to unmodified, re-purposed residential rooms, with everything in between. Yet recordings built entirely within a laptop-and-headphones production space have no problem sitting in shuffled playlists next to work from high end, world-class studios. Acoustic features—or lack thereof—are variously leveraged and overcome through recording craft and signal processing to adapt almost any production space to the sonic needs of the recording artist.

Session 5aNS

Noise and ASA Committee on Standards: Louis C. Sutherland's Lifetime Contributions to the Fields of Noise, Standardization, and Classroom Acoustics

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

David Lubman, Cochair
DL Acoustics, 14301 Middletown Ln., Westminster, CA 92683-4514

Chair's Introduction—8:30

Invited Papers

8:35

5aNS1. My collaboration with Louis C. Sutherland on classroom acoustic issues. David Lubman (DL Acoust., 14301 Middletown Ln., Westminster, CA 92683-4514, dlubman@dlacoustics.com)

8:55

5aNS2. Lou Sutherland—Bridging acoustics and people in acoustics. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 101789, Germany, b.schulte-fortkamp@tu-berlin.de)

Lou Sutherland's scientific contributions have helped to renew the acoustic world. In all the years of my participation at ASA meetings, Lou has provided valuable insights that have advanced many subfields of noise. These include aircraft noise reduction, atmospheric sound propagation, classroom learning, and soundscape. Lou is always open to new ideas to improve acoustical calculations and measurements. His extensive work in classroom acoustics with David Lubman has increased his prominence in noise and architectural acoustics. For example, classroom acoustics has become one of the most persistent topics on the agendas of the Technical Committees on Noise and Architectural Acoustics for at least a decade. Although Lou and I have never worked together his publications on aircraft noise and soundscape have strongly influenced my research. Lou has an endearing and effective way of "inducting" new people into the ASA world. I remember with gratitude how Lou stimulated me to become active in ASA's Noise and Architecture Acoustics communities by introducing me to its distinguished members. I am very pleased for this opportunity to thank Lou for his past work. And to continue our valuable discussions that make the world a better place by reducing noise and improving soundscape.

9:15

5aNS3. Lou Sutherland—Contributions in aircraft noise. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooksacoustics.com)

Lou Sutherland has made pivotal contributions to the understanding of aircraft noise and its effects and to developing solutions for those problems. His work in noise effects ranges from the 707 aircraft prototype, the development of an engine noise suppressor for the B-52 to reduce sonic structural fatigue, to the Saturn rocket program. Lou's interests have encompassed aircraft engine turbomachinery cascade acoustic resonances and the propagation of sound through the atmosphere. These contributions were made while Lou was a researcher first at Boeing and later at Wyle Labs, as Chief Scientist and Deputy Director of Research. His insights have led to creative solutions to reduce aircraft noise effects on structures and on humans. He served as President of INCE and as an Associate Editor of JASA. In 2002 Lou was awarded the Acoustical Society of America Silver Medal in Noise "For contributions to the solution of aerospace and community noise problems, and for studies of molecular absorption and classroom acoustics." Most importantly, Lou has a terrific, infectious enthusiasm for all that he does, which is combined with a warm, generous spirit. Lou has provided kind and gentle leadership and guidance to many grateful beneficiaries over the years.

9:35

5aNS4. Lou Sutherland: A friend and colleague over the years. Paul D. Schomer (Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

I have known Lou for a short time compared to some of you, only 50 years. Over these years, our paths have crossed in many ways at many times: reviewing the US Army's research program that I set up in 1972, development of structural response of buildings and its application to impulse noise and aircraft noise, joining with me and others in a collaborative consultative effort near the St. Louis airport, interacting with me as associate editor of JASA, and participating in Standards—notably the classroom acoustics standard.

9:55–10:15 Break

10:15

5aNS5. Lou Sutherland—The man and his work. Ben H. Sharp (7802 Trammell Rd., Annandale, VA 22003, bhs940@yahoo.com)

Lou Sutherland is one of the longest serving members of the international acoustics community and one of the most prolific. In a career lasting over 60 years, his main areas of research have involved studies of human and structural response to noise, sonic boom and blast, but his contributions to the fields of acoustics and vibration have been much more diverse and have included topics such as atmospheric absorption, aircraft noise measurement and modeling, low-frequency noise, and acoustic testing. He has been active in standards organizations and as a result of his services has been elected as a Fellow in several professional societies. As a consultant, Lou had the ability to break down complex problems into simple components that could be understood and solved, often using limited available data and simple theoretical models. Lou is still active as a consultant and continues to influence people with his ideas. As a long-term colleague, the author will examine some of his major contributions and share some of his experiences working with Lou.

10:35

5aNS6. Tribute to Lou Sutherland. Robert Kull (Marstel-Day, LLC, 203 Norton St., San Antonio, TX 78260, rkull@marstel-day.com)

This will be a brief tribute to Lou Sutherland who I first met in 1989. Although Lou Sutherland recognized that I had no acoustics background back then, he treated me with utmost respect and professionalism. Lou mentored, guided, and challenged me while I was program director for the Noise and Sonic Boom Impact Technology Advanced Development Program Office for the Air Force's Armstrong Medical Research Laboratory. I sought his wisdom as I faced critical decisions for my Service's research program. Over these past 25 years, we haven't had many occasions to work together, but the times we have had were significant and I will always cherish them.

10:55

5aNS7. Understanding speech in noisy rooms: Continuing the Sutherland legacy. Peggy B. Nelson (Univ. of Minnesota, 164 Pillsbury Dr. Se, Minneapolis, MN 55455, peggynelson@umn.edu)

Lou Sutherland has inspired us to continue to study the problems listeners experience when listening to signals in noisy rooms. We continue to follow his lead to learn more about the variability that adult and child listeners experience when listening in noise. Our current work allows adult listeners with hearing loss to self-adjust hearing aids in noisy rooms. Results suggest that hearing aid users vary significantly in the gain that they self-select for listening in noise. In general, as noise increases, most, but not all, listeners select less hearing aid gain. Overall, individual listeners varied greatly in their gain preferences even when hearing losses were very similar. These results suggest that we have much to learn about the processes of listening in complex environments. Variables will be discussed. The implications of this work suggest that a great deal is left unknown about performance in background noise. [Work supported by NIDCD R01 DC013267.]

11:15

5aNS8. Absorption of sound at high altitude: Lou Sutherland's contribution. Richard Raspet (NCPA, Univ. of MS, National Ctr. for Physical Acoust., University of MS, University, MS 38677, raspet@olemiss.edu), Andi Petculescu (Phys., Univ. of Louisiana Lafayette, Lafayette, MS), and Oleg A. Godin (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado, Boulder, CO)

Lou Sutherland was one of the original developers of ANSI S1-26, "Method for the calculation of the absorption of sound in the atmosphere." Research in the propagation of infrasound in support of the Comprehensive Test Ban Treaty Organization demonstrated the need for a reformulation of the calculation in the stratosphere and thermosphere. Lou and Hank Bass developed the formulation appropriate to the composition and environment of the upper atmosphere. The paper [J. Acoust. Soc. Am. **115**(3), 1012–1032 (2004)] documenting this work is an impressive example of the compilation of databases and ideas into a coherent model to predict the absorption of sound. In conclusion, we will briefly review recent research extending the analysis of infrasound in the upper atmosphere.

Session 5aSC

Speech Communication: Speech Perception and Production in Noise and Related to Disorders of Speech, Language or Hearing (Poster Session)

Noah H. Silbert, Chair
Communication Sciences & Disorders, University of Cincinnati, 3202 Eden Avenue, 344 French East Building, Cincinnati, OH 45267

Posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to view other posters, contributors of odd-numbered papers will be at their posters from 8:15 a.m. to 9:45 a.m., and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 11:30 a.m. There will be a 15-minute break from 9:45 a.m. to 10:00 a.m.

Contributed Papers

5aSC1. Affective prosody production in autistic and typically developed adult males. Daniel J. Hubbard, Daniel J. Faso, Noah J. Sasson, and Peter F. Assmann (School of Behavioral and Brain Sci., GR4.1, Univ. of Texas at Dallas, P.O. Box 830688, Richardson, TX 75083, dhubbard@utdallas.edu)

This study examined differences in production of affective prosody in adult males with autism spectrum disorder (ASD) and typically developed (TD) controls. Previous studies of children with ASD have reported increased variability in fundamental frequency (f0) in spontaneous and semi-spontaneous speech compared to TD children. A controlled set of expressive speech recordings was collected from 30 talkers (15 ASD) to measure differences between the two groups using the same lexical content. Isolated vowels, vowel-consonant-vowel (VCV) syllables, words and short phrases were elicited in five emotion contexts: angry, happy, interested, sad, and neutral. The recordings were obtained using evoked and portrayed elicitation techniques: talkers were asked to recall past emotional episodes (evoked) and role-play scripted scenarios (portrayed) specific to each emotion context. Consistent with previous work and extending the findings to adults producing the same lexical content, talkers with ASD showed increased f0 variability in each emotion context except for neutral. In addition, systematic group differences were found in acoustic properties other than f0 used to convey affective prosody—including harmonics-to-noise ratio and intensity—which were higher in each emotion context for talkers with ASD compared to TD talkers.

5aSC2. Production of contrastive focus in children with autistic spectrum disorder. lucile Rapin, Paméla Trudeau-Fisette, Marie Bellavance-Courtemanche, and Lucie Ménard (Linguist, UQAM, C.P. 8888, Succursale Centre-Ville, Montreal, Quebec H3C 3P8, Canada, lucilerapin@gmail.com)

Contrastive focus serves to emphasize the importance of a semantic unit in the language string. Children with autistic spectrum disorder (ASD) appear to show difficulties in producing this prosodic marker. This study aimed to identify acoustic correlates related to contrastive focus in children with ASD. Nine francophone children with ASD and nine francophone typically developing (TYP) children produced simple four-word sentences (for example, «C'est une chaise.»: «it is a chair.») in a neutral condition and then in a contrastive focus condition. Ninety-six speech productions were recorded using a system that synchronized acoustic signals with lingual and labial movements. Maximum pitch, mean pitch, and pitch range, as well as maximum and mean sound intensity and duration were investigated. Values for pitch range, maximum and mean sound intensity, and duration were greater in the focus condition than in the neutral condition. Moreover, the differences were significantly greater in TYP children than in ASD children, who did not have increased speech values when switching to the focus mode. This suggests that pitch range, and intensity and duration of sound correlate most with contrastive focus marking in both groups. Yet, it appears that ASD children show less contrastive focus marking than TYP children.

5aSC3. Fundamental frequency of speech directed to children who have hearing loss. Mark VanDam, Paul De Palma, and William E. Strong (Speech & Hearing Sci., Washington State Univ., PO Box 1495, Spokane, WA 99202, mark.vandam@wsu.edu)

Studies of child-directed speech (CDS) have shown that when talking to children, parents systematically use (among other strategies) increased fundamental frequency (F_0). Lombard effects such as increased F_0 have also been documented when addressing a listener who is hard-of-hearing (HH). Here, we examine F_0 of mothers and fathers in families with HH versus typically developing (TD) children in CDS and adult-directed speech (ADS) contexts. Whole-day audio recordings were collected by a child-worn audio recorder and analyzed by automatic speech recognition (ASR) software to identify segments of vocal activity by children and their parents (LENA Research Foundation, Boulder, CO). Custom software extracted F_0 values in all conditions. We found that (1) mothers are much more systematic in their use of CDS than fathers, (2) parents do not appear to be sensitive to the hearing status of their children, and (3) parents of HH children may have higher overall F_0 irrespective of CDS or ADS. Results suggest that mothers and fathers do not use F_0 with their children in the same way, and parents of children with hearing loss may have certain global F_0 characteristics not shared by parents of TD children.

5aSC4. Action verb processing correlates with motor asymmetry in Parkinson's disease. Emily Wang, Lee K. Walters (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 1611 West Harrison St., Ste. 530, Chicago, IL 60612, emily_wang@rush.edu), and Leo A. Verhagen Metman (Neurology, Rush Univ. Medical Ctr., Chicago, IL)

Onset of motor symptoms in PD is asymmetrical, and symptoms affect the side of onset more severely throughout the progression of the disease, which is known as "motor asymmetry." Previous studies revealed that PD patients with right-motor asymmetry (more severe symptoms on the right side of body) exhibited more speech impairment than those with left-motor asymmetry (Wang *et al.*, 2003; 2006). Patients with active deep brain stimulation (DBS) exhibited more severe speech deficits when they received either bilateral or left-only stimulation of the subthalamic nucleus (STN) than receiving right-only STN stimulation (Santens *et al.*, 2003). These findings lead to the current hypothesis that there is correlation between the motor asymmetry and linguistic processing of action verbs. Using a novel action verbal processing paradigm, 24 PD patients (12 right- and 12 left-motor asymmetry) and 11 age- gender-matched healthy controls were tested. The results showed that when the motor asymmetry was sufficiently different, i.e., when the difference of the UPDRS scores between the two sides of the body is greater than 2 (points), the right-motor asymmetry patients took

longer to process the action verbs, supporting the hypothesis that motor asymmetry is correlated with linguistic processing of action verbs at the cortical level.

5aSC5. Tongue movement pattern in speakers with dysarthria in production of "Ohio". Jimin Lee (Commun. Sci. and Disord., The Penn State Univ., 404A Ford Bldg., University Park, PA 16802, jxl91@psu.edu)

A different range of tongue motion during speech production has been assumed in speakers with dysarthria across different severity groups based on the previous findings in acoustic vowel space. To further confirm these findings, this study examined the tongue movement pattern in speakers with dysarthria during the production of "Ohio," a target word that requires the consecutive production of multiple vowels. Ten speakers with dysarthria of varying severity participated in this study. Three dimensional electromagnetic articulography (Wave system) was utilized to identify tongue movement trajectory during the target word production. Each speaker produced three repetitions of the word "Ohio" in the carrier phrase "I say a__ again." X, y, and z coordinate values from the tongue sensor (positioned approximately 25 mm from the tongue apex) were recorded. Statistical analyses showed that speakers with severe dysarthria produced a) less overall tongue movement, b) slower speed, c) less absolute tongue advancement-retraction and tongue height movement, and d) greater distance between starting /o/ and ending /o/ positions than speakers with mild dysarthria. Results are discussed with regard to the interpretation of tongue movement displacement and trajectory shape difference in speakers with dysarthria across different severity groups.

5aSC6. A comparative study of variability in landmark sequences and implications for dysphonic speech analysis. Keiko Ishikawa (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, 5371 Farmridge Way, Mason, OH 45040, ishikak@mail.uc.edu), Marepalli B. Rao (Dept. of Environ. Health, Univ. of Cincinnati, College of Medicine, Cincinnati, OH), and Suzanne Boyce (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

Dysphonia negatively affects speech intelligibility especially in the presence of background noise. This may be because dysphonic speech often contains a higher proportion of noise and/or a lower proportion of harmonic power, leading to reduced information in the speech signal. Landmark (LM) analysis was designed to identify patterns of information in the speech signal that are particularly salient for the auditory system. Consequently, it describes speech as a sequence of LMs. Past studies successfully differentiated disordered speech from normal speech based on the number of times each LM occurs. While the count was a sufficient measure for their purposes, transitional patterns in LM sequences may yield more descriptive information on underlying mechanism of the intelligibility deficits. Shannon's Entropy and Markov chain model were used to evaluate the difference in LM sequences between normal and dysphonic speech. Landmarks were obtained from the first sentence of the Rainbow Passage for 33 normal speakers and 36 dysphonic speakers using SpeechMark^{TM} software package. The variability in the transitional patterns of LM was significantly less in dysphonic speech compared to normal speech. This suggests intelligibility deficits may be due to the greater acoustical constraints inherent to dysphonic speech.

5aSC7. The effect of Parkinson disease on voice onset time: Temporal differences in voicing contrast. Jason A. Whitfield (Speech Pathol. & Audiol., Kent State Univ., A104 Kent Ctr. for Performing Arts, Kent, OH 44242, jwhitfi4@kent.edu) and Alexander M. Goberman (Commun. Sci. and Disord., Bowling Green State Univ., Bowling Green, OH)

Parkinson disease (PD) affects the basal ganglia, which is involved with the selection, sequencing, and implementation of movement. Investigations suggest an extension of deficits to the speech motor and linguistic systems. Previous studies examining voice onset time (VOT) suggest that VOTs are neither systematically delayed nor systematically advanced in PD as compared to controls. The VOTs for voiced and voiceless stops are expected to differ for both individuals with PD and controls. However, inefficiencies in the sequencing and implementation of the voicing gesture may result in a smaller difference between VOT values of voiced and voiceless cognates. The current study examined VOT in individuals with PD and controls. Participants produced a corpus of stimuli to evaluate VOT using the carrier phrase "CVp again." The four corner vowels and all stop consonants were used. Speech was recorded and VOT values were manually measured for every utterance. Overall VOTs were significantly shorter for voiceless stops across all speakers. However, the average difference between the voice and voiceless cognates produced by the PD group was significantly smaller than the control group. These data suggest that the PD group exhibited less temporal distinction between voiced and voiceless stops.

5aSC8. Syllable position effects on perceptual determination of /r/ errors in expert and naive listeners. Sarah M. Hamilton, Keiko Ishikawa, Hedieh Hashemi Hosseinabad, Suzanne Boyce, and Lindsay Mullins (Commun. Sci. and Disord., Univ. of Cincinnati, 3433 Clifton Ave., Cincinnati, OH 45220, hamilsm@mail.uc.edu)

When ordinary listeners hear words containing a phoneme whose production differs in one feature from the target, the anomaly is often ignored if the word is recognizable. Phonetic research and clinical practice both depend on phonetics training to mitigate these top-down listening effects and enable detection of anomalies. Children with speech sound errors for American English /r/ are believed to have better productions when /r/ occurs at the beginning of a word. However, the shorter duration of /r/ word-initially may be leading clinicians to under-identify errors. While previous studies have shown that clinicians are better than naive listeners at identifying error /r/ in single syllables, they have not assessed identification of /r/ errors in whole words, where position may interact with lexical bias effects. In this study, speech-language pathologists and naive listeners rated children's natural speech productions of word-initial and word-final/r/ in whole words. These /r/ productions in words were selected to make a continuum defined by normalized third formant values. Reaction times and responses to stimuli were compared between listener groups. Initial results indicate that clinicians and naive listeners differ in their responses to formant profiles of stimuli, but that syllable position exerts a strong effect for both groups.

5aSC9. The use of visual information in non-native speech sound discrimination across the first year of life. D. Kyle Danielson, Padmapriya A. Kandhadai, and Janet F. Werker (Psych., Univ. of Br. Columbia, 2136 West Mall, Vancouver, British Columbia V6T 1Z4, Canada, kdanielson@psych.ubc.ca)

Infants are able to match seen and heard speech even in non-native languages, and familiarization to audiovisual speech appears to affect subsequent auditory-only discrimination of non-native speech sounds (Danielson *et al.*, 2013; 2014). However, the robustness of these behaviors appears to change rapidly within the first year of life. In this current set of studies, conducted with six-, nine-, and 10-month-old English-learning infants, we examine the developmental trajectory of audiovisual speech perception of non-native speech sounds. In the first place, we show that the tendency to detect a mismatch between heard and seen speech sounds in a non-native language changes across this short period in development, in tandem with the trajectory of auditory perceptual narrowing (Werker & Tees, 1984; Kuhl *et al.*, 1992; *inter alia*). Furthermore, we demonstrate that infants' familiarization to matching and mismatching audiovisual speech affects their auditory speech perception differently at various ages. While six-month-old infants' auditory speech perception appears to be malleable in the face of prior audiovisual familiarization, this tendency declines with age. The current set of studies is one of the first to utilize traditional looking-time measurements while also employing pupillometry as a correlate of infants' acoustic change detection (Hochmann & Papeo, 2014).

5aSC10. Effects of reading ability on lexically-informed perceptual learning. Emily Thompson, Stephen Graham, Alexandra T. Bohner, Julia R. Drouin, and Rachel M. Theodore (Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT 06269-1085, emily.thompson@uconn.edu)

Research on perceptual learning for speech shows that lexical information can be used to modify phonological representations. Moreover, lexically informed perceptual learning is domain general in that it also

influences the mapping to phonology from printed text. Given phonological processing deficits in adults with reading impairment, we hypothesized that reading ability would meditate how lexical information is used to dynamically adjust perceptual representations. Adult participants were identified as either good or poor readers based on standardized assessments of reading and reading sub-skills. All completed a lexical decision training phase followed by a test phase. Stimuli in both phases consisted of printed text. During training, readers viewed an ambiguous grapheme midway between "N" and "H." Lexical information was used to bias perception of the grapheme as either "N" or "H." At test, all readers categorized tokens along an "N" to "H" continuum. The results to date indicate that both groups of readers used lexical information to modify letter perception; however, the learning effect was more robust in the good readers compared to the poor readers. These results suggest that the degree to which lexical information is used to modify the mapping to sound structure is related to phonological ability.

5aSC11. Effect of intelligibility and speech rate on perceived listener effort. Kathleen Nagle (Speech & Hearing Sci., MGH Dept, of Laryngeal Surgery, Univ. of Washington, One Bowdoin Square, 11th Fl., Boston, MA 02114, kfnagle@uw.edu)

Perceived listener effort (PLE) has been defined as the perceived "amount of mental exertion required to attend to, and understand, an auditory system" (McGarrigle *et al.*, 2014). PLE may be affected by multiple characteristics of the speech signal, including sentence length and intelligibility. Previous qualitative data indicates that listeners are also sensitive to speaking rate. This experiment measured the effect of speech rate on PLE for speech produced with a range of intelligibility. Eleven listeners with normal hearing (HINT), working memory (WAIS-IV), and receptive vocabulary (PPVT) transcribed and rated their PLE for the 144 low-context sentences produced with a monotone electrolarynx (EL) set at 75 Hz. Speech rate (syllables per second) was measured from onset of the first word to offset of the last word. Multiple regression analysis indicated that intelligibility and speech rate accounted for a significant amount of variance in PLE scores, $R^2 = 0.70$, $F(2,142) = 168.803$, $p < 0.001$, $R^2_{adj} = 0.70$. Intelligibility had a unique negative effect on PLE ratings ($b = -6.45$, $SE = 0.35$), $t(142) = -18.37$, $p < 0.001$, $sr^2 = 0.70$; however, rate was not uniquely predictive ($p > 0.05$). Future research should examine the association between radiated noise and PLE for EL speech.

5aSC12. Elevated depressive symptoms associate with an emotion-general deficit in speech perception at a cocktail party. Zilong Xie (Dept. of Commun. Sci. & Disord., The Univ. of Texas at Austin, 2504A Whitis Ave. (A1100), Austin, TX 78712, xzilong@gmail.com), W. Todd Maddox (Dept. of Psych., The Univ. of Texas at Austin, Austin, TX), and Bharath Chandrasekaran (Dept. of Commun. Sci. & Disord., The Univ. of Texas at Austin, Austin, TX)

Individuals with major depressive disorder demonstrate significant deficits in social communication. While the majority of work on individuals with elevated depressive symptoms focuses on speech production, much less is known about speech perception in individuals with elevated depressive symptoms. Here, we examine speech perception in the presence of different types of distractions. Two forms of distraction are hypothesized to interfere with speech perception in such listening conditions: energetic masking (EM) and information masking (IM). Relative to EM, IM places greater demands on executive function. A recent study showed that individuals with elevated depressive symptoms exhibit a selective deficit in perception of neutral speech during primarily IM conditions, likely due to their impairments in executive function. Here, we examined whether this selective deficit extends to emotional speech that portrays anger, fear, happiness, or sadness. Results showed that during IM conditions, individuals with elevated depressive symptoms exhibited poorer performance for neutral as well as for the four emotion categories, compared with those without elevated depression symptoms. However, both groups performed comparably during EM conditions. These findings suggest that elevated depressive symptoms are associated with an emotion-general deficit in speech perception in noise conditions that place demands on executive function.

5aSC13. An investigation into how the acoustics of different sized open plan classrooms affects speech perception in Kindergarten children. Kiri T. Mealings, Katherine Demuth (Linguist, Macquarie Univ., Level 3 AHH Macquarie Univ., Sydney, New South Wales 2936, Australia, kiri.mealings@students.mq.edu.au), Jörg Buchholz, and Harvey Dillon (National Acoust. Labs., Sydney, New South Wales, Australia)

Open plan classrooms, where several classes share the same space, have recently re-emerged in Australian primary schools. This paper examined how the acoustics of four Kindergarten classrooms (an enclosed classroom (25 students), a double classroom (44 students), a linear fully open plan triple classroom (91 students), and a semi-open plan K-6 classroom (205 students)) affect speech perception. Twenty-two to 23 children in each classroom participated in an online four-picture choice speech perception task while adjacent classes engaged in quiet versus noisy activities. The noise levels recorded during the task were higher in the larger open plan classrooms compared to the smaller classrooms for both the quiet and noisy conditions. A linear mixed effects model revealed that children's performance accuracy decreased as noise level increased. Additionally, children's speech perception abilities decreased the further away they were seated from the loudspeaker, and this effect was stronger the higher the noise level. Children's response time was also slower in the noisiest compared to quietest classroom. These results suggest that open plan classrooms may not be appropriate learning environments for didactic-style teaching with young children due to their high intrusive noise levels which negatively impact speech perception.

5aSC14. Using the stability of vocal onsets to evaluate vocal effort in response to changing acoustical conditions. Mark L. Berardi (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, markberardi12@gmail.com), Eric J. Hunter (Dept. of Commun. Sci. and Disord., Michigan State Univ., East Lansing, MI), and Timothy W. Leishman (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Several acoustical measures have been used in the past to evaluate vocal effort. They are useful in evaluating occupational risk for teachers or other occupational voice users. Analysis of vocal onsets has been used to show vocal effort in spasmodic dysphonia. Acoustical measures are also used in clinical speech-language pathology as an inexpensive and noninvasive ways of evaluating pathology severity and tracking therapy progress. In this presentation, a single acoustic parameter based on relative fundamental frequencies of glottal pulses following voiceless consonants (the onset coefficient) is used to evaluate vocal effort in response to changes in background noise and reverberation time within speaking environments. Analysis shows that males and females have similar vocal effort levels in the most typical acoustical conditions. However, females respond to louder background noise and longer reverberation times with more vocal effort than males.

5aSC15. Vocal comfort and effort in speech: Accommodation to different room acoustic conditions. Simone Graetzer, Eric J. Hunter, and Pasquale Bottalico (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, sgraetz@msu.edu)

Background: Vocal effort is a physiological entity that accounts for variation in voice production as loading increases, measured as sound pressure level (SPL). A number of studies have investigated vocal effort and load, but few have considered the role of acoustic clarity (C50), the early to late arriving sound ratio. Method: 20 subjects performed vocal tasks in various room acoustic conditions. Tasks were performed in three styles: soft style, comfortable conversational style, and loud classroom style. C50 in the position of the talker was changed by means of two reflective panels. Two noise levels were used: background noise, and artificial child babble noise. After each task, the subject answered questions addressing their perception of vocal comfort, control and fatigue. SPL and Fundamental Frequency (F0) were measured. Results: When panels were present, talkers perceived the room as being more comfortable to speak in. In particular, in the babble condition and in loud speech, comfort and control tended to increase. Vocal effort (SPL) tended to decrease when panels were present. An assessment of F0 is also reported. The results indicate that even while keeping reverberation time constant, reflective surfaces may be optimized to increase voice comfort and reduce vocal effort.

5aSC16. Vocal fatigue over a workday: A schoolteacher case study. Michael K. Rollins, Mark L. Berardi (Dept. of Phys. and Astronomy, Brigham Young Univ., 1371 N 380 W, Provo, UT 84604, michael.rollins@byu.edu), Eric J. Hunter (Dept. of Communicative Sci. & Disord., Michigan State Univ., East Lansing, MI), and Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

Due to the vocal demands in their workplace and other factors, schoolteachers appear to be especially susceptible to long-term voice problems. Because female schoolteachers appear to engender an even higher risk, this case study focused on speech measures of one female elementary schoolteacher who reported feeling hoarse. The teacher was recorded over the course of a typical school day. Certain speech traits in the teacher's speech performance during elicitation tasks were evaluated by comparing changes from the beginning to the end of the school day. A strong correlation between the mean, standard deviation, skew, and kurtosis of long term average spectra (LTAS) and vocal fatigue was found, while other measures were found to be less consistent. The methods and findings will be discussed, and areas for further research suggested.

5aSC17. Vowel and sibilant sound production in noise. Kevin J. Reilly (Audiol. & Speech Pathol., Univ. of Tennessee, 12633 Wagon Wheel Circle, Knoxville, TN 37934, kreilly3@uthsc.edu)

The present study investigated speech production in noise and whether speakers modulate the spectral contrasts of vowel and sibilant sounds depending on the frequency characteristics of the background noise signal. Twelve speakers participated in the present study. Speakers were presented with three-word sequences one word at a time on a computer monitor and, after a variable delay, produced the three words in the order they were presented. Word sequences were comprised of CVC words containing either or both of the vowels, /a/ and /ae/, and either or both of the sibilants, /s/ and /ʃ/ , in word-initial position. During the presentation and production of each sequence, speakers were exposed to one of four possible noise conditions: (1) silence; (2) vowel masking; (3) sibilant masking; and (4) speech-shaped noise. The sibilant masker contained energy at frequencies associated with a speaker's productions of /s/ and /ʃ/ and the vowel masker at frequencies associated with their productions of /a/ and /ae/. These masking signals were generated prior to the experiment based on practice productions of the stimulus words by each speaker. Spectral contrast distances are based on vowel formant frequencies and sibilant spectral moments (1–4). Noise condition effects on spectral contrast distances are being evaluated to determine whether speakers selectively alter their production of those sounds masked by background noise.

5aSC18. Perceived emotional valence in clear and conversational speech for older adults with hearing loss. Shae D. Morgan and Sarah H. Ferguson (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Ste. 1201, Salt Lake City, UT 84112, shae.morgan@utah.edu)

When asked to speak as though talking to an individual with hearing loss, talkers modify their speech from their everyday conversational style to a "clear" speaking style. Audiologists have observed in practice that individuals with hearing loss sometimes complain that their frequent communication partners are shouting at them, while the communication partners insist they are just trying to speak more clearly. A previous study investigated how individuals with normal hearing perceived emotional valence in clear and conversational speech and found that clear speech sounded angry more often than conversational speech. The present study will explore whether a similar effect is found in older individuals with hearing loss. Perceived anger has been partially attributed to increased energy in the high frequencies, where age-related hearing loss is most severe. Furthermore, aging effects have been found for visual emotion perception. Older adult listeners with hearing loss will be presented with conversational and clear sentences from the Ferguson Clear Speech Database (Ferguson, 2004) and asked to assign an emotion category to each sentence (anger, sadness, happiness, fear, disgust, or neutral). The results will indicate whether elderly listeners with hearing loss judge clear speech as sounding angry more often than typical conversational speech.

5aSC19. Speaking rate effects on phonemic boundary perception in cochlear implant users. Brittany N. Jaekel, Rochelle S. Newman, and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland College Park, LeFrak Hall, College Park, MD 20742, jaekel@umd.edu)

Rate of speech can affect the durations of certain phonemes (Crystal & House, 1982), and listeners must normalize for speech rate to properly identify words (Miller, 1981). How speech rate affects perception of phonemic boundaries in users of cochlear implants (CIs) in comparison to normal-hearing (NH) listeners is unknown. The speech processing that occurs in these devices may obscure word and phoneme boundaries, providing less reliable durational information about the incoming speech signal. Less effective rate normalization in CI users could thus contribute to the degraded speech perception often observed in this population. Preliminary results with stop-consonant series show absolute differences between NH and CI listeners in the precise location of phonemic boundaries (with CI users' boundaries typically occurring at shorter voice onset time durations), but not in the relative relationship of these boundaries as a function of speech rate. Data from NH listeners presented a CI simulation show effects of number of channels on both the location of phonemic boundaries and presence of rate normalization.

5aSC20. Effect of context on bimodal benefit for temporally interrupted sentences in simulated electric-acoustic stimulation. Soo Hee Oh, Gail Donaldson (Univ. of South Florida, Tampa, FL), and Ying-Yee Kong (Northeastern Univ., 226 Forsyth Bldg., 360 Huntington Ave., Boston, MA 02115, yykong@neu.edu)

Previous research has demonstrated a clear effect of linguistic context on bimodal benefit for continuous speech, implicating an interaction between low-frequency acoustic cues and top-down linguistic processing (Kong *et al.*, *CIAP 2013*). However, bimodal benefit appears to be reduced for phonemic restoration (Baskent, 2012 *JARO*), suggesting that low-frequency facilitation of top-down processing is weakened when the speech stream is interrupted. The present study examined top-down effects in temporally interrupted speech by comparing bimodal benefit for low- and high-context sentences. Young, normal-hearing listeners were presented with City University of New York (CUNY) or Institute of Electrical and Electronics and Engineers (IEEE) sentences that were gated with silence (5 Hz, 50% duty cycle). One ear received sentences that were noise-band vocoded (8, 12, or 16 channels for CUNY; 12, 16, or 32 channels for IEEE) and the other ear received low-pass (LP) speech or LP harmonic complexes (LPHCs). Findings demonstrated clear effects of context on bimodal benefit when LP speech was presented to the residual-hearing ear, however, the benefits observed were considerably smaller than those reported previously for continuous speech. Unlike previous findings for continuous speech, no bimodal benefits were observed when LPHCs were presented to the LP ear.

5aSC21. Cutaneous vibration enhances the Lombard Effect. François-Xavier Brajot (Commun. Sci. and Disord., Ohio Univ., 1 Ohio University Dr., Grover W221, Athens, OH 45701, brajot@ohio.edu) and Vincent L. Gracco (Commun. Sci. and Disord., McGill Univ., Montreal, Quebec, Canada)

The Lombard Effect is traditionally defined as a compensatory vocal response to noise at the ear. Somatosensory feedback is also relevant to speech, however, and may be expected to play a role in the Lombard response and perceived vocal effort. The effect of masking both auditory and laryngeal vibrotactile feedback was assessed in adults without speech or hearing disorders. Autophonic loudness, speech intensity, fundamental frequency, and formant frequencies were measured in each of three conditions: unmasked, auditory masking, and mixed auditory plus somatosensory masking. Neither the slope of the autophonic loudness curve nor the shape of the oral reading intensity contour changed as a function of masking condition.

However, the intercept of the autophonic loudness curve and the offset of the intensity contour were greater in the mixed masking as compared to the auditory masking condition, indicating that vibrotactile masking effectively enhanced the Lombard Effect. These findings support the hypothesis that both auditory and somatosensory feedback function to adjust speech gain, without affecting relative production intensity or autophonic loudness perception. As such, the Lombard Effect is proposed to result from a general sensorimotor process rather than from a specific audio-vocal mechanism.

5aSC22. Segmental interference by the masker modulation spectrum in sentences: Effects of age and hearing loss. Daniel Fogerty (Dept. of Commun. Sci. and Disord., Univ. of South Carolina, University of South Carolina, Keenan Bldg., Ste. 300, Columbia, SC 29208, fogerty@sc.edu), Jayne B. Ahlstrom, William J. Bologna, and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Younger normal-hearing, older normal-hearing, and older hearing-impaired subjects listened to sentences with consonants or vowels replaced with noise modulated by the envelope of a competing talker. The modulation spectrum of the noise was low-pass filtered at different cutoff frequencies. Sentences were spectrally shaped according to listeners' individual audiometric thresholds to ensure sufficient audibility; in addition, a second group of younger normal-hearing subjects listened to sentences that were spectrally shaped according to the mean audiogram of the hearing-impaired subjects. Results demonstrated declines in sentence intelligibility for older as compared to younger listeners, with more evident declines for older hearing-impaired listeners. Declines in sentence intelligibility based on vowel cues with increasing modulation rates in the competing noise were noted for all listener groups. In contrast, sentence intelligibility based on consonant cues increased when the competing noise included modulation rates between 8 and 16 Hz. An adverse effect of spectral shaping was also observed, with an overall decrease in performance for younger spectrally matched listeners. Thus, in temporally complex listening conditions, spectral shaping to assure audibility may not fully control for reduced sentence intelligibility among hearing-impaired listeners and may interact with temporal properties of the speech signal. [Work supported by NIH/NIDCD and ASHA.]

5aSC23. Phonemic restoration with envelope and periodicity cues: Effects of age and competing talkers. William J. Bologna (Dept. of Hearing and Speech Sci., Univ. of Maryland, 135 Rutledge Ave., MSC 550, Charleston, SC 29425, bologna@musc.edu), Monita Chatterjee (Auditory Prostheses & Percept. Lab., Boys Town National Res. Hospital, Omaha, NE), and Judy R. Dubno (Dept. of Otolaryngol. - Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Perception of interrupted speech improves when noise is inserted in silent gaps between speech segments. This "phonemic restoration" is enhanced with intervening noise that contains low-level speech information, such as noise modulated by the temporal envelope of the missing speech. However, this benefit may be reduced in multiple-talker environments where competing speech contributes additional masking effects. To minimize these masking effects, periodicity information within intervening noise segments of interrupted speech may help listeners segregate multiple voices and thus increase phonemic restoration. The relative benefit of envelope and periodicity cues, and their use by older adults with poorer performance in multiple-talker environments, remains unclear. To address these questions, younger and older adults with normal hearing listened to target sentences in quiet and competing talker backgrounds; target sentences were periodically interrupted with silence, envelope-modulated noise, or pulse trains, which contained periodicity information from the missing speech. Phonemic restoration was defined as the difference in recognition of interrupted sentences filled with envelope-modulated noise or pulse trains as compared to sentences interrupted by silence. Results are discussed in terms of contributions of envelope and periodicity cues to perceptual organization in complex listening environments. [Work supported by NIH/NIDCD and a AAA Student Investigator Research Grant.]

5aSC24. Across-formant integration and speech intelligibility: Effects of acoustic source properties in the presence and absence of a contralateral interferer. Robert J. Summers, Brian Roberts (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, r.j.summers@aston.ac.uk), and Peter J. Bailey (Dept. of Psych., Univ. of York, York, United Kingdom)

Three experiments used three-formant (F1 + F2 + F3) analogues of natural sentences to explore the role of acoustic source properties in across-formant integration. In experiment 1, F1 + F3 were generated using a monotonous periodic source (F0 = 140 Hz) and second-order resonators (H1 + H3); in experiment 2, F1 + F3 were tonal analogues (T1 + T3). F2 could take either form (H2 or T2). Target formants were always presented monaurally; the target ear was assigned randomly on each trial. In some conditions, only the target was present; in others, a competitor for F2 (F2C) was added contralaterally. Listeners must reject F2C to optimize recognition. Competitors (H2C or T2C) were created using the time-reversed frequency and amplitude contours of F2. Without F2C, intelligibility was reasonably high and the effect of a source mismatch between F1 + F3 and F2 was negligible. For both shared- and hybrid-source targets, the impact of adding F2C was modest when it was tonal but large when it was harmonic, irrespective of whether F2C matched F1 + F3. Experiment 3 showed that this pattern was maintained when corresponding harmonic and tonal analogues were loudness-matched. These findings extend those from earlier research using dichotic targets. Source type and competition, rather than acoustic similarity, govern the phonetic contribution of a formant. [Work supported by ESRC.]

5aSC25. Informational masking of monaural speech by a single contralateral formant. Brian Roberts and Robert J. Summers (Psych., School of Life and Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, b.roberts@aston.ac.uk)

Recent research suggests that the ability of an extraneous formant to impair intelligibility depends on the variation of its frequency contour. This idea was explored using a method that ensures interference occurs only through informational masking. Three-formant analogues of sentences were synthesized using a monotonous periodic source (F0 = 140 Hz). Target formants were presented monaurally; the target ear was assigned randomly on each trial. A competitor for F2 (F2C) was presented contralaterally; listeners must reject F2C to optimize recognition. In experiment 1, F2Cs with various frequency and amplitude contours were used. F2Cs with time-varying frequency contours were effective competitors; constant-frequency F2Cs had far less impact. Amplitude contour also influenced competitor impact; this effect was additive. In experiment 2, F2Cs were created by inverting the F2 frequency contour about its geometric mean and varying its depth of variation over a range from constant to twice the original (0–200%). The impact on intelligibility was least for constant F2Cs and increased up to ~100% depth, but little thereafter. The effect of an extraneous formant depends primarily on its frequency contour; interference increases as the depth of variation is increased until the range exceeds that typical for F2 in natural speech. [Work supported by ESRC.]

5aSC26. Effect of noise on foreign-accent adaptation. Elisa Ferracane, Cynthia P. Blanco, Gabriela Cook, Karen Johnson, and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, 305 E. 23rd St., Austin, TX 78712, elisa@ferracane.com)

Previous work indicates that listeners are able to adapt quickly to changes in accent [Clarke & Garrett, 2004; Bradlow & Bent, 2008] and that adaptation is faster for familiar accents [Witteman *et al*., 2013; Blanco *et al*., 2014]. The present study examines how listeners adapt to more and less familiar accents presented in noise as compared to quiet, since noisy listening environments are more perceptually challenging [Vaden *et al*., 2013; Gilbert *et al*., 2014] but also more closely represent typical listening conditions. Native English listeners heard blocks of sentences produced by native- or foreign-accented talkers (Korean, Spanish) and responded to a visual probe. Listener accuracy and reaction times were compared across accent and speaker blocks. Results showed that, unlike in quiet, listeners were unable to adapt to the less-familiar Korean accent. Processing of the familiar Spanish accent was successful in noise and quiet. Adaptation in noise was furthermore correlated with talker intelligibility: less intelligible talkers elicited lower accuracy and slower response; in quiet, there was no talker effect. The results revealed that the ability to adapt to a novel foreign accent is disrupted, as the extra perceptual effort in noise prevents listeners from generalizing about systematic variability of foreign-accented features.

5aSC27. Gap detection and speech recognition in noise in younger versus older listeners. Susan M. DeMetropolis, Janet R. Schoepflin, and Lawrence J. Raphael (Commun. Sci. and Disord., Adelphi Univ., 158 Cambridge Ave., Garden City, NY 11530, susandemetropolis@mail.adelphi.edu)

The ability to encode information in temporal envelopes of acoustic stimuli is an important skill of the auditory system. It has been demonstrated that a listener's temporal-processing capability is predictive of performance on speech recognition, especially in noisy and complex environments (George, Festen, & Houtgast, 2006; George *et al.*, 2007; Shen, 2014; Snell, Mapes, Hickman, & Frisina, 2002). The goal of this study was to determine whether temporal acuity using gap detection thresholds from the Gap-in Noise (GIN) test (Musiek, 2003) has any relation to the perception of speech in noise from the Revised Speech Perception in Noise (R-SPIN; Bilger, 1984). All younger listeners were between 21 and 30 years (n = 7/mean = 24.7) and older listeners (n = 4/mean = 67.25) were between 65 and 71 years old with audiometric thresholds equal or better than 15 dB HL between 250 and 8000 Hz in both ears, monolingual English, and intact cognitive skills. Results revealed that temporal acuity of older listeners was reduced compared to the younger listeners. Despite temporal processing differences between the two age groups on the GIN test, they performed similarly on the R-SPIN. Results suggest that impaired abilities to process envelope and fine structure cues of acoustic signals does not influence speech perception in background noise.

FRIDAY MORNING, 22 MAY 2015 KINGS 1, 8:30 A.M. TO 11:00 A.M.

Session 5aSP

Signal Processing in Acoustics: Detection, Classification, and Analysis

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Contributed Papers

8:30

5aSP1. Performance analysis of a fuzzy soft decision CFAR detector in non-Rayleigh background. Yanwei Xu, Chaohuan Hou, Shefeng Yan, and Xiaochuan Ma (IOA CAS, Beisihuanxilu NO.21, Beijing 100191, China, xyw@mail.ioa.ac.cn)

A fuzzy statistical normalization fuzzy constant false alarm rate (FSN-FCFAR) detector in non-Rayleigh background based on fuzzy statistical normalization and fuzzy soft decision is proposed. The performance of the proposed fuzzy soft decision detector is studied both for homogeneous backgrounds and for non-homogeneous environments caused by interfering targets or clutter edges. Performance comparisons with the conventional hard decision CFAR detectors such as CA-CFAR, GO-CFAR and OS-CFAR are carried out. The comparison results show that the proposed FSN-FCFAR detector can not only get a very good detection performance in homogeneous backgrounds, but also can confront interfering targets and clutter edges at the same time in non-homogeneous environments. Moreover, the fuzzy soft decision detector can provide more valuable information than the hard decision detector for data fusion, target tracking or object identification.

8:45

5aSP2. Lucky ranging in underwater acoustic environment subject to spatial coherence loss. Hongya Ge (Elec. & Comput. Eng., New Jersey Inst. of Technol., University Heights, Newark, NJ 07102, ge@njit.edu) and Ivars P. Kirsteins (Naval Undersea Warfare Ctr., Newport, RI)

In principle, a passive sonar system utilizing a long line array can estimate a source's range by measuring its wavefront curvature. However, implementation can be difficult because wavefront curvature ranging is highly sensitive to spatial coherence losses. Real-world data measured by towed arrays often exhibit rapid fluctuations in signal spatial coherence across time and frequency. Such non-stationary degradation in spatial coherence on distributed arrays bears the similarity to the atmospheric Kolmogorov Turbulence Effect present in ground based telescopes (blurring of stars). Inspired by the lucky imaging approach used in astronomy, we propose a new approach for passive ranging in underwater environments subject to time-varying spatial coherence loss. Over short time scales, we speculate that the ocean may be well-behaved in the sense that there are little or no distortion to the signal wavefronts aka lucky moments. If detected, these lucky moments can be utilized to estimate range more accurately, overcoming coherence loss degradations experienced over long integration times. Modeling the signal as either coherent or incoherent in a short time frame, we derive the maximum likelihood lucky range estimator and show that with proper time-scale and processing bands, lucky moments can be identified that yield greatly improved estimates of range.

9:00

5aSP3. Classification of signals in spherically invariant random clutter. Bruce K. Newhall and Anna Slowikowski (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723, bruce.newhall@jhuapl.edu)

The generalized likelihood ratio test (GLRT) is derived for the case of a signal subspace in acoustic clutter characterized by a spherically invariant random variable (SIRV). This result is a generalization of two previous results. First, the GLRT for the detection of a one dimensional signal in

SIRV clutter has been previously given. However, featureless classification work has previously considered signals that are members of a multidimensional subspace but only in Gaussian clutter. The SIRV model extends that to the non-Gaussian clutter case. A SIRV is the product of two random variables: a non-Gaussian scalar times a complex multivariate Gaussian vector. The general GLRT result is then applied to a generalized gamma distribution for the SIRV scalar. When the generalized gamma exponent parameter is -2 this produces a product SIRV whose acoustic intensity has a generalized Pareto distribution. When the exponent parameter is $+2$, the SIRV intensity is k distributed. This demonstrates that two widely used clutter distribution models are special cases of this more general distribution. Methods of parameter estimation for application of the technique are also given. [Work supported by the Office of Naval Research.]

9:15

5aSP4. An approximation method for dispersive wave propagation. Jonathan Ben-Benjamin, Leon Cohen (Dept. of Phys., Hunter College, 695 Park Ave., New York, NY, yonatan@greatwing.com), and Patrick Loughlin (BioEng., Univ. of Pittsburgh, Pittsburgh, PA)

A phase space approximation method that extends a previous single-mode approximation [Loughlin, Cohen, J. Acoust. Soc. Am. **118**, 1268 (2005)] for linear dispersive wave propagation is developed. We show that each mode is governed by a Schrodinger-type equation, where the corresponding Hamiltonian operator is non-Hermitian if the dispersion relation is complex, for which case there is absorption. The propagated wave is obtained by evolving each mode according to its respective Schrodinger equation. We show how to obtain the initial modes from the initial conditions on the wave. We then formulate the propagation problem in phase space and obtain the exact equation of motion for the phase space function. We also obtain an approximate solution for the phase space evolution of the wave, which involves a simple substitution into the initial phase space function. Examples are given for a parallel plate wave guide and the beam equation. [Work supported by ONR, code 321US.]

9:30

5aSP5. Environmentally corrected matched filter. H. John Camin (Graduate Program in Acoust., Penn State Univ., ARL Penn State Univ., P.O. Box 30, M.S. 9310L, State College, PA 16804, jcamin@psu.edu)

Matched filtering is commonly used to process broadband active acoustic echoes providing higher levels of range resolution and reverberation noise suppression than can be realized through narrowband processing. Since theoretical processing gains are proportional to the signal bandwidth, it is typically desirable to utilize the widest band signals possible. However, as signal bandwidth increases, so do environmental effects that increase decorrelation between the received echo and the transmitted waveform. This loss of coherence often results in processing gains and range resolution much lower than theoretically predicted. Weiner filtering, commonly used in image processing to improve distorted and noisy photos, was investigated as an approach to correct for these environmental effects. This improved signal processing, environmentally corrected matched filter (ECMF), first uses a Wiener filter to estimate the environmental transfer function and then again to correct the received signal. This process can be viewed as a "smarter" inverse or whitening filter that optimally adjusts according to the signal to noise ratio across the spectrum. Sonar simulation toolset (SST) synthetic data and measured in-air data will be presented to illustrate the improved processing.

9:45

5aSP6. Three component vibrational time reversal communication. Brian E. Anderson, Timothy J. Ulrich, and James A. Ten Cate (Geophys. Group (EES-17), Los Alamos National Lab., MS D446, Los Alamos, NM 87545, bea@lanl.gov)

Time reversal provides an optimal prefilter matched signal to apply to a communication signal before signal transmission. Time reversal allows compensation for wave speed dispersion and can function well in reverberant environments. Time reversal can be used to focus elastic energy to each of the three components of motion independently. A pipe encased in concrete was used to demonstrate the ability to conduct communications of information using three component time reversal. The ability of time reversal to compensate for multi-path distortion (overcoming reverberation) will be demonstrated and the rate of signal communication will be presented. [The U.S. Department of Energy, through the LANL/LDRD Program, is gratefully acknowledged for supporting this work.]

10:00

5aSP7. Analysis of signal detection SNR limits in snapshot-deficient scenarios with colored noise. Jose A. Diaz-Santos (Naval Surface Warfare Ctr., 18444 Frontage Rd., Ste. 323, Dahlgren, VA 22448-5161, jose.diaz-santos@navy.mil) and Kathleen E. Wage (George Mason Univ., Fairfax, VA)

Most source enumeration algorithms assume a white noise background; however, in many underwater applications, the background noise is colored. Various researchers have investigated the problem of source enumeration when the background noise is colored. The standard approach is to apply a whitening filter before the signal enumeration algorithm. Most of these algorithms perform poorly when the number of noise snapshots used to estimate the whitening filter is small. Nadakuditi and Silverstein [IEEE J. Sel. Topics Signal Process., 2010] developed an algorithm for source enumeration using random matrix theory that provides the fundamental SNR limits for snapshot-deficient scenarios with arbitrary noise. Nadakuditi and Silverstein's analysis focuses on the performance when the number of signal plus noise snapshots varies while the number of snapshots used for the whitening filter stays constant. This talk analyzes the performance of this algorithm when number of snapshots available to estimate the whitening filter varies from N to 10 N. Simulations using two different colored noise models for a large vertical linear array located in the deep ocean will be presented.

10:15

5aSP8. Effect of reverberation on instrumental vibrato tones. Sarah R. Smith and Mark F. Bocko (Dept. of Elec. and Comput. Eng., Univ. of Rochester, 405 Comput. Studies Bldg., P.O. Box 270231, Rochester, NY 14627, sarahsmith@rochester.edu)

Musical vibrato is the process by which a performer subtly varies the frequency and amplitude of a note for expressive effect. Previous work has suggested that the vibrato modulation may not be uniform across all overtones, causing the individual frequency trajectories to become decorrelated. In this study, we investigate the effect of reverberation on instrumental tones performed with vibrato. This is accomplished by performing a detailed time-frequency analysis to extract the instantaneous frequencies of many overtones. In order to accurately track the low amplitudes of the upper harmonics, we present an algorithm tailored for optimal tracking in as little as 10 dB SNR. For tones recorded in an anechoic environment, the instantaneous frequency modulations of all the overtones are found to be highly correlated. However, tones recorded in a reverberant space, as well as anechoic recordings processed with a synthetic reverberation are found to have significantly lower correlations between the overtones. In addition to providing insight into musical timbre variations, these results could be useful for characterization of acoustic transfer functions or applications where overtone correlation is assumed, such as audio source separation.

10:30

5aSP9. Time delay estimation of acoustic tones with unpredictable frequency variations under low signal to noise ratio conditions. Stephen J. Franklin, Anthony Finn, and Joshua Meade (School of Eng., Univ. of South Australia, W2-56 Mawson Lakes Campus, UniSA, Mawson Lakes, South Australia 5095, Australia, stephen.franklin@unisa.edu.au)

This paper describes a novel approach for autonomously detecting and tracking aircraft in the vicinity of a unmanned aerial vehicle (UAV). The time difference of arrival (TDOA) of acoustic tones emanating from the distant aircraft are correlated between spatially distributed microphone pairs located on the UAV. The geometry of multiple microphone pairs then allows the elevation and azimuth of the approaching aircraft to be estimated, despite the high levels of onboard narrow- and broadband noise emanating from the engine firing sequence, propeller, airflow over the microphones, and mechanical vibration. Current signal processing techniques estimate time delay when the signal level of the approaching aircraft is above the background signature of the sensing aircraft, despite subtle frequency variations in the narrowband components of the approaching aircraft's signature.

The technique described in this paper is designed to track these unpredictable frequency variations and thus extends the detection range of the approach by enabling estimation of TDOA when the signal level is 20 dB below the noise floor. Potential detection ranges in excess of 1 km are demonstrated when the signal processing is combined with careful suppression of the many noise sources onboard the UAV.

10:45

5aSP10. A novel high-resolution algorithm for separating ray paths interrupted by colorful noise. Longyu Jiang (Southeast Univ., Sipailou 2#, Nanjing 210096, China, JLY01412@gmail.com)

Abstract: In the first step of ocean acoustic tomography, ray paths need to be identified with parameters which will be utilized in inversion process. Many second-order direction finding algorithms have been applied to identify ray paths in this context, such as beamforming, MUSIC algorithm. Because ray paths are produced by reflection and (or) refraction of emitted signal, they are fully correlated or coherent. Recently, the smoothing-MUSICAL algorithm, which is based on second-order statistics, has been developed to separate fully correlated or coherent signals and gained large improvement of separation ability in DOA-temporal domain. All the above methods work under the assumption of white Gaussian noise, however, in the real ocean environment there always exists colorful noise. In this paper, we propose a high-resolution algorithm based on four-order cumulants, which enables to separate fully correlated or coherent signals interrupted by colorful noise. Its performance has been illustrated by synthetic experiments and real experiments.

FRIDAY MORNING, 22 MAY 2015 — BALLROOM 4, 8:00 A.M. TO 11:30 A.M.

Session 5aUW

Underwater Acoustics: Propagation, Tomography, Scattering, and Transducers

Joseph D. Schneiderwind, Cochair
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Contributed Papers

8:00

5aUW1. Striation processing of continuously active sonar data. Scott Schecklman and Lisa M. Zurk (Elec. and Comput. Eng., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, sscheck@pdx.edu)

Previous work by the authors has shown that striations associated with the waveguide invariant can provide a new physics-based tracking constraint that improves tracking performance in pulsed active sonar data. A necessary preprocessing step is to isolate the target return from the direct arrival and other target echoes. Striation-based beamforming can be used as an optional preprocessing step to preserve the multipath structure in the beamformer output of a horizontal line array. Recent interest in continuously active sonar (CAS) processing has motivated research to determine how striations might be similarly exploited for tracking when the sonar is operated with a high duty cycle. In CAS signals, the direct arrival is almost always overlapping in time with the target echo return and the standard time-gating technique cannot isolate the echo return. In this paper, a time-dependent filter is introduced to isolate and extract the CAS echo spectrum. The conditions in which striation-based beamforming should be applied to preserve the striation structure in the beam output are also explored. The physics-based signal processing algorithms discussed here are expected to provide valuable information about the target's multipath structure, which can be exploited in an extended Kalman filter for enhanced tracker performance.

8:15

5aUW2. Comparisons between energy-flux propagation models utilizing single- and multiple-scattering treatments of sea-surface roughness in a range-independent waveguide. Joseph D. Schneiderwind, Derek Olson, Charles W. Holland, and Anthony Lyons (Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, jds563@psu.edu)

Acoustic scattering from a rough sea surface in propagation modeling is generally treated using a single-scatter approximation in which only specular components are considered and the incoherent field is neglected. This work presents energy-flux calculations of transmission loss for a shallow water waveguide with a rough sea-surface following a Pierson-Moskowitz spectrum. Multiple scattering is evoked by considering conservation of energy from scattering events. Comparisons are made between the model with multiple scattering and that using the single-scatter approximation. Models are compared to the direct solution using the boundary element method. [This work was partially funded by the Department of Defense (DoD) through the National Defense Science & Engineering Graduate Fellowship (NDSEG) Program, the Eric Walker Fellowship from the Applied Research Laboratory at the Pennsylvania State University, and the Achievement Rewards for College Scientists (ARCS) Foundation Pittsburgh Chapter.]

8:30

5aUW3. A follow up low frequency propagation experiment in Currituck Sound. Richard D. Costley (GeoTech. and Structures Lab., U.S. Army Engineer Res. & Development Ctr., 3909 Halls Ferry Rd., Vicksburg, MS 39180, dan.costley@usace.army.mil), Andrew R. McNeese, Megan S. Ballard, Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas, Austin, TX), Kent K. Hathaway (Coastal and Hydraulics Lab., US Army Engineer Res. & Development, Kitty Hawk, NC), Eric Smth (GeoTech. and Structures Lab., U.S. Army Engineer Res. & Development Ctr., Vicksburg, MS), Preston S. Wilson, and Thomas G. Muir (Appl. Res. Labs., The Univ. of Texas, Austin, TX)

An initial low frequency propagation experiment was conducted in September 2013 in which a Combustive Sound Source (CSS) generated transient acoustic signals in water depths of approximately 2.5 m [Costley *et al.* J. Acoust. Soc. Am. **136**, 2178 (2014)]. High amplitude signals were generated mid-water column and recorded with bottom mounted hydrophones at distances up to 200 m. A time harmonic, two-dimensional axisymmetric finite element model under-predicted the magnitudes of the recorded pressure signals. A follow up experiment was conducted in October 2014. In addition to using the CSS and the bottom mounted hydrophones, a 4-element vertical line array was deployed. Measurements were made at source-receiver separations from approximately 10 m to 1000 m in 100 m increments. Additionally, acoustic properties of the sediment were obtained through in-situ measurements [McNeese *et al.* J. Acoust. Soc. Am. **136**, 2252 (2014)]. The results of the experiment, along with results of the finite element model using updated sediment properties, will be presented. [Work supported by the U.S. Army Engineer Research and Development Center. Permission to publish was granted by Director, Geotechnical & Structures Laboratory.]

8:45

5aUW4. Efficient estimation of the probability density function of transmission loss in uncertain ocean environments. Brandon Patterson and David R. Dowling (Univ. of Michigan, 626 Spring St., Apt. #1, Ann Arbor, MI 48103-3200, awesome@umich.edu)

Predictions of acoustic transmission loss (TL) in the ocean are useful in a variety of naval and ocean engineering applications. However, real ocean environments are imperfectly known and have many uncertainties. Consequently, TL calculations performed for these environments are also uncertain. Traditional Monte-Carlo methods used to obtain probability density functions (PDF) of TL in such circumstances require many field calculations, and are impractical for real-time applications. This presentation describes how the PDF of TL at a point of interest may be estimated using a single TL-field calculation and the statistics of TL within a range-depth area containing that point. To document the performance of this technique, RAM was used to perform TL field calculations for 100 Hz and 300 Hz sound sources in realistic ocean sound channels with eight uncertain environmental parameters describing the bathymetry, sound speed profile, and bottom properties. Preliminary results indicate that TL PDFs generated with area statistics have an L_1-error $\leq$ 0.5 when compared with 1000-calculation Monte-Carlo PDF results throughout most of the sound channel. However, greater errors were observed in the near field as well as in areas where the sound channel depth was less than 100 m, for 100 Hz sound. [Sponsored by ONR.]

9:00

5aUW5. Scattering of plane acoustic waves at elastic particles with rough surfaces. Leif Bjorno (Stendiget 19, Taastrup 2630, Denmark, prof.lb@mail.dk)

A comprehensive theoretical and numerical study of the influence of surface roughness of elastic particles in water on the scattering of ultrasonic waves has been carried out. For near spherical shape of the particles and with small rms-roughness heights, a perturbation method has been developed. In this method, the first-order perturbation contribution predicts the contribution to the incoherently scattered acoustic field due to surface roughness, and the second-order perturbation contribution predicts the change in the coherent field and will satisfy the requirement of energy conservation. The second-order perturbation contribution is evaluated by use of the form function concept, while the first-order perturbation to the total scattered acoustic field is evaluated by use of the scattering cross-section. As a function of the ka-value and for different rms-roughness heights a numerical study of the forward and backward scattering from rough, elastic particles has been carried out and a substantial roughness influence on the scattered field has been verified. Some experimental results from measurements of scattering from glass and cast iron spheres have given evidence to the numerical predictions.

9:15

5aUW6. Measured backscattering of a first order vortex beam by a sphere with helicity selective processing and imaging. Viktor Bollen, Daniel S. Plotnick, David J. Zartman, and Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, viktor.bollen@wsu.edu)

An acoustic vortex beam's wavefield has a null on the axis of propagation and an angular phase ramp. A four-element transducer generates a first order vortex beam in water by appropriately phasing each element [V. Bollen *et al.*, Proc. Meet. Acoust. **19**, 070075 (2013)]. A differential beam having a broader null may also be produced by exciting two elements out-of-phase. The four-element transducer was also used to measure backscattering by a small solid sphere scanned in a raster pattern and insonified by the vortex beam. By independently recording each of the four elements and adding appropriate time delays in post-processing, the scattering may be measured in a helicity neutral, and co- and cross-helicity sensitive modes; this allows the retention of the vortex source without modifying the experimental setup. Due to the rapid phase change on the axis of the beam, subwavelength resolution of the null location of the sphere can be achieved. By applying a time delay-and-sum imaging algorithm this resolution can be further increased. Three-dimensional images can also be constructed from the data. By driving two opposing elements, the tilt of the transducer with respect to the scanning plane can be inferred, aiding alignment. [Work supported by ONR.]

9:30

5aUW7. Bubble nonlinear oscillation in sound field. Desen Yang, Shiyuan Jin, Shengguo Shi, Jie Shi, and Haoyang Zhang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No.145 Nantong St., Nangang District, Heilongjiang, Harbin 150001, China, jinshiyuan.seaky@163.com)

The complicated mechanism of gas bubble oscillation in sound field is quite fascinating. One simplified case is a single bubble motivated by specific finite-amplitude sound waves. Based on the modified Keller-Miksis model, a theoretical and numerical investigation of the regular and nonlinear radial oscillations of a gas bubble driven by different given finite-amplitude waves is presented. Some crucial control parameters of the bubble radial oscillations, including radius of gas bubble, acoustic frequency and acoustic pressure, are studied with multiple numerical analysis methods. Parameter regions which are benefit for power spectrum variation of the excitation forces are given, which can provide the foundation for potential engineering applications.

9:45–10:00 Break

10:00

5aUW8. Modes of plates and shells in water driven by modulated radiation pressure of focused ultrasound. Timothy D. Daniel, Phil L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164, timothy.daniel@email.wsu.edu), Ahmad T. Abawi (HLS Res., La Jolla, CA), and Ivars Kirsteins (NUWC, Newport, RI)

We used modulated radiation pressure of focused ultrasound to excite resonant flexural vibrations of an aluminum circular plate in water. The source transducer was driven with a double-sideband suppressed carrier voltage as previously used for exciting low-frequency modes of liquid drops [P. L. Marston and R. E. Apfel, J. Acoust. Soc. Am. **67**, 27–37 (1980)]. The response of the target (detected with a hydrophone) was at twice the modulation frequency and proportional to the square of the drive voltage. Due to the spatially localized nature of the radiation pressure of the focused beam, mode shapes could be identified by scanning the source along the target while measuring the target's response. Additional measurements were done with an open-ended water-filled copper circular cylindrical shell in which resonant frequencies and mode shapes were also identified. These experiments illustrate how high-frequency focused sound can be used to identify relatively low-frequency modes of elastic objects without direct contact. [Work supported by ONR.]

10:15

5aUW9. Multi-fluid cavity underwater acoustic transducer. Yongjie Sang and Yu Lan (College of Underwater Acoust. Eng., Harbin Eng. Univ., 145 Nantong St., Nangang District, Harbin, Heilongjiang Province 150001, China, sangyongjie@126.com)

A multi- fluid cavity underwater acoustic transducer is developed. The main structures of the transducer include inner Helmholtz resonator, Janus longitudinal vibration transducer and two outer Helmholtz resonators. The vibration of Janus transducer at low frequencies is enlarged by the inner Helmholtz resonator, and the vibration of Janus transducer at high frequencies is enlarged by the two outer Helmholtz resonators, so this kind of transducer can transmit broadband acoustic signal in water. Operation bandwidth of the multi-fluid cavity underwater acoustic transducer is usually more than two octaves. The broadband operating mechanism of the transducer is studied. The effect of some of the major structural parameters on the operation bandwidth and Helmholtz resonance frequency is studied by finite element software ANSYS. The structure size of virtual prototype is obtained. A prototype is produced based on finite element design results, the electro-acoustic performance are tested in anechoic tank. The results agree well with the simulation results. Operation bandwidth of the prototype is 500 Hz–2500 Hz, the maximum source level greater than 200 dB.

10:30

5aUW10. Metal electrode direct sound generation in seawater. Michael S. McBeth (SPAWAR Systems Ctr. Atlantic, SSC Atlantic/NASA Langley Res. Ctr., 11 West Taylor St., M.S. 207, Hampton, VA 23681, m.s.mcbeth@ieee.org)

Thermoacoustic expansion of water due to Ohmic heating and volume changes in the very thin Helmholtz double layer that forms adjacent to metal electrodes are two mechanisms that have been used to explain sound generation by metal electrodes in seawater. Using 20 gauge solid sterling silver wire electrodes with only the circular cross section of the wire exposed, our experiments have demonstrated intense sound production in the form of acoustic tone bursts in seawater from a thermoacoustic mechanism at a second harmonic of 20 kHz of the electrical driving frequency at 10 kHz. Additionally, we observed a separate intense and transient acoustic tone burst generation from a mechanism we attribute to vesicle layer excitation where the vesicle is a highly charged layer of water associated with the boundary of droplets and bubbles. This vesicle layer excitation sound generation mechanism is distinct from any Helmholtz double layer mechanism since the double layer is always present while the acoustic tone burst we observe repeatedly occurs after about 4.4 ms of applied alternating voltage at 10 kHz and last only about 560 μs. We present our experimental setup and results. Moving seawater directly to generate sound may lead to more compact projectors.

10:45

5aUW11. A comparison between flexible disk transducer cavitation measurements and predictions. Arthur Horbach and James McEachern (Navmar Appl. Sci. Corp., 65 West St. Rd., Warminster, PA 18974, horbach@navmar.com)

An investigation of the threshold of cavitation of flexible disk transducers used as underwater acoustic sources was conducted. The principal task was the estimation of the cavitation threshold in sound pressure level (SPL) of an individual transducer as a function of operating depth and the actual measurement of the cavitation threshold. The transducers under test were driven at a series of voltage levels, using CW pulse lengths of 50 and 100 ms. these drive voltages were increased in steps and the received signals were analyzed for total harmonic distortion (THD), transmitting voltage response (TVR), output SPL, and voltage-current phase angle, all as a function of depth. The onset of cavitation on the flex disk is well defined in terms of THD. Comparing the measured values for the onset of cavitation to those predicted by an expression recommended by Sherman and Butler, a value of the cavitation parameter, gamma, was determined to be about 0.36, consistent with, and bracketed by, their values, for a number of transducer shapes. predictions of cavitation threshold versus depth for the flexible disk transducer were within 1 to 2 dB of measured results.

11:00

5aUW12. Spatial structure of temporal coherence scale under the different propagation conditions. Linhui Peng, Jianhui Lu, and Xiaotao Yu (Ocean Technol., Information College, Ocean Univ. of China, 238 Songling Rd., Qingdao, Shangdong 266100, China, penglh@ouc.edu.cn)

Internal wave induce the decreasing of temporal coherence of sound field. The scale of temporal coherence is related with strength of the internal wave and frequency of the sound wave; meanwhile, it has the definite spatial structure. Spatial structures of the temporal coherence are not same under the different propagation conditions. It is related with the structure of the sound field.

11:15

5aUW13. Experimental study of ultra shallow water acoustic wave propagation. Konstantin Dmitriev, Alisa Dorofeeva, and Sergei Sergeev (Phys., Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, sergeev@aesc.msu.ru)

The sound propagation in ultra shallow water was studied experimentally. The investigation was carried out in the nature pond approximately 300×20 m in size. Its depth was about 1 m. For such a pond critical frequency is about 700 Hz that is easy to emit and receive. The main purpose was to study the process of acoustic wave propagation in case of strong bottom influence and to restore some bottom parameters. The signal was radiated by the transducer and recorded by a set of point receivers located at different distances from the transducer. The signal was linear frequency modulated in 100 Hz–10 kHz band and was 10 s in duration. Its correlation with the recorded signal defines the pond response function. The modes representation was used. The group velocity frequency dependence was obtained by use of this method and it was very close to the theoretical one considering a soft bottom. The critical frequency also corresponded to the soft bottom model. The attenuation of sound was also studied in a wide frequency range including non-propagating mode. The results are very sensitive to the signal-to-noise ratio. It was shown that the used correlation method significantly improved the quality of obtained characteristics.

Poster paper 5aUW14 will be on display for the entire session.

5aUW14. Simulated joint reconstruction of shallow water features using acoustic tomography methods. Sergei Sergeev, Andrey Shurup, and Alisa Scherbina (Phys., Moscow State Univ., Leninskie Gory, Moscow 119991, Russian Federation, sergeev@aesc.msu.ru)

The possibility of simultaneous reconstruction of different shallow water parameters by the mode tomography methods is considered. The initial data for the reconstruction are the propagation times of individual modes in the different frequency bands. The reconstruction results can be both the water layer parameters (such as sound speed profile, eddies) and the bottom characteristics (such as relief and sound speed in the bottom). The approach is based on idea that the different shallow water parameters affect the propagation times of different modes on the different frequencies in the different manner. As a result, if it is possible to separate from the received total field enough information about the propagation times of the mode signals, then it becomes possible to implement the joint reconstruction of the considered shallow water features. Another peculiarity of the regarded approach that all the considered shallow water features are described using the single basis. As a result, the single perturbation matrix is used for the joint reconstruction. The numerically simulated results obtained by the regarded approach are presented.

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Information for contributors to the Journal of the Acoustical Society of America (JASA)

Editorial Staff[a)]

Journal of the Acoustical Society of America, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300

The procedures for submitting manuscripts to the *Journal of the Acoustical Society of America* are described. The text manuscript, the individual figures, and an optional cover letter are each uploaded as separate files to the *Journal's* Manuscript Submission and Peer Review System. The required format for the text manuscript is intended so that it will be easily interpreted and copy-edited during the production editing process. Various detailed policies and rules that will produce the desired format are described, and a general guide to the preferred style for the writing of papers for the *Journal* is given. Criteria used by the editors in deciding whether or not a given paper should be published are summarized.

PACS numbers: 43.05.Gv

TABLE OF CONTENTS

I. INTRODUCTION

The present document is intended to serve jointly as (i) a set of directions that authors should follow when submitting articles to the *Journal of the Acoustical Society of America* and as (ii) a style manual that describes those stylistic features that

a)E-mail: jasa@aip.org

are desired for the submitted manuscript. Authors may refer to recent issues of the *Journal* for examples of how specific style issues are handled.

II. ONLINE HANDLING OF MANUSCRIPTS

All new manuscripts intended for possible publication in the *Journal of the Acoustical Society of America* should be submitted by an online procedure. The steps involved in the processing of manuscripts that lead from the initial submission through the peer review process to the transmittal of an accepted manuscript to the production editing office are handled by a computerized system referred to here as the Peer X-Press (PXP) system. The Acoustical Society of America contracts with AIP Publishing LLC for the use of this system. There is one implementation that is used for most of the material that is submitted to the *Journal of the Acoustical Society of America* (JASA) and a separate implementation for the special section *JASA Express Letters (JASA-EL)* of the *Journal.*

A. Registration

Everyone involved in the handling of manuscripts in the *Journal's* editorial process must first register with the *Journal's* implementation of the PXP system, and the undertaking of separate actions, such as the submission of a manuscript, requires that one first log-in to the system at http://jasa.peerx-press.org/cgi-bin/main.plex.

If you have never logged into the system, you will need to get a user name and password. Many ASA members are already in the data base, so if you are a member, you in principle may already have a user name and password, but you will have to find out what they are. On the login page, you click on the item "Unknown/Forgotten Password." On the new page that comes up after you do this, give your first name and last name. After you have filled in this information, just click on "mailit." You will then get a e-mail message with the subject line "FORGOTTEN PASSWORD." The system will actually give you a new password if you had ever used the system before. After you get this new password, you can change it to something easy to remember after you login.

Once you have your "user name" and "password" you go to the log-in page again, and give this information when you log-in. You will first be asked to change your password. After you do this, a "task page" will appear. At the bottom of the page there will be an item *Modify Profile/Password.* Click on this. Then a Page will appear with the heading *Will you please take a minute to update the profile?*

If you are satisfied with your profile and password, then you go to the top of the Task page and click on the item *Submit Manuscript* that appears under *Author Tasks.* Then you will see a page titled *Manuscript Submission Instructions.* Read what is there and then click *continue* at the bottom of the page.

B. Overview of the editorial process

(1) An author denoted as the corresponding author submits a manuscript for publication in the *Journal.*
(2) One of the *Journal's* Associate Editors is recruited to handle the peer-review process for the manuscript.
(3) The Associate Editor recruits reviewers for the manuscript via the online system.
(4) The reviewers critique the manuscript, and submit their comments online via the Peer X-Press system.
(5) The Associate Editor makes a decision regarding the manuscript, and then composes online an appropriate decision letter, which may include segments of the reviews, and which may include attachments.
(6) The *Journal's* staff transmits a letter composed by the Associate Editor to the corresponding author. This letter describes the decision and further actions that can be taken.

If revisions to the manuscript are invited, the author may resubmit a revised manuscript, and the process cycle is repeated. To submit a revision authors should use the link provided in the decision message.

C. Preparation for online submission

Before one begins the process of submitting a manuscript online, one should first read the document *Ethical Principles of the Acoustical Society of America for Research Involving Human and Non-Human Animals in Research and Publishing and Presentations* which is reached from the site <http://scitation.aip.org/content/asa/journal/jasa/info/authors>. During the submission, you will be asked if your research conformed to the stated ethical principles and if your submission of the manuscript is in accord with the ethical principles that the *Acoustical Society* has set for its journals. If you cannot confirm that your manuscript and the research reported are in accord with these principles, then you should not submit your manuscript.

Another document that you should first read is the document *Transfer of Copyright Agreement,* which is downloadable from the same site. When you submit your manuscript online you will be asked to certify that you and your coauthors agree to the terms set forth in that document. What is in that document has been carefully worded with extensive legal advice and which has been arrived at after extensive discussion within the various relevant administrative committees of the Acoustical Society of America. It is regarded as a very liberal document in terms of the rights that are allowed to the authors. One should also note the clause: The author(s) agree that, insofar as they are permitted to transfer copyright, all copies of the article or abstract shall include a copyright notice in the ASA's name. (The word "permitted" means permitted by law at the time of the submission.) The terms of the copyright agreement are non-negotiable. The Acoustical Society does not have the resources or legal assistance to negotiate for exceptions for individual papers, so please do not ask for such special considerations. Please read the document carefully and decide whether you can provide an electronic signature (clicking on an appropriate check box) to this agreement. If you do not believe that you can in good conscience give such an electronic signature, then you should not submit your manuscript.

Given that one has met the ethical criteria and agreed to the terms of the copyright transfer agreement, and that one has decided to submit a manuscript, one should first gather together the various items of information that will be requested during the process, and also gather together various files that one will have to upload.

Information that will be entered into the PeerX-Press submission form and files to be uploaded include:

(1) Data for each of the authors:
  (a) First name, middle initial, and last name
  (b) E-mail address
  (c) Work telephone number
  (d) Work fax number
  (e) Postal address (required for corresponding author, otherwise optional)

(2) Title and running title of the paper. The running title is used as the footline on each page of the article. (The title is limited to 17 words and the running title is limited to six words and up to 50 characters and spaces; neither may include any acronyms or any words explicitly touting novelty.)

(3) Abstract of the paper. (This must be in the form of a single paragraph and is limited to 200 words for regular articles and to 100 words for letters to the editor. (Authors would ordinarily do an electronic pasting from a text file of their manuscript.)

(4) Principal ASA-PACS number that characterizes the subject matter of the paper and which will be used to determine the section of the *Journal* in which the published paper will be placed. Note that if the PACS number you list first is too generic, e.g., 43.20, that may result in a delay in processing your paper.

(5) A short prioritized list of Associate Editors suggested for the handling of the manuscript.

(6) Contact information (name, e-mail address, and institution) of suggested reviewers (if any), and/or names of reviewers to exclude and reasons why.

(7) Cover letter file (optional, with some exceptions). Material that would ordinarily have been in the cover letter is now supplied by answering online questions and by filling out the online form. However, if an author needs to supply additional information that should be brought to the attention of the editor(s) and/or reviewer(s), a cover letter should be written and put into the form of an electronic file.

(8) Properly prepared manuscript/article file in LaTeX, Word, or WordPerfect format. (The requirements for a properly prepared manuscript are given further below.) It is also possible to submit your file in PDF but this is not desirable since the entire manuscript must be retyped. It must be a single stand-alone file. If the author wishes to submit a LaTeX file, the references should be included in the file, not in a separate BibTeX file. Authors should take care to insure that the submitted manuscript/article file is of reasonable length, no more than 2 MB.

(9) Properly prepared figure files in TIFF, PS, JPEG, or EPS (see also, Section V. H); one file for each cited figure number. The uploading of figures in PDF format is not allowed. (The captions should be omitted, and these will appear as a list in the manuscript itself.) The figures should not have the figure numbers included on the figures in the files as such, and it is the responsibility of the corresponding author to see that the files are uploaded in proper order. Authors may upload figures in a zip file (figure files must be numbered in order using 1, 2, etc. If figures have parts they must be numbered 1a, 1b, 1c, etc.). [In order to maintain online color as a free service to authors, the *Journal* cannot accept multiple versions of the same file. Authors may not submit two versions of the same illustration (e. g., one for color and one for black & white). When preparing illustrations that will appear in color in the online *Journal* and in black & white in the printed *Journal,* authors must ensure that: (i) colors chosen will reproduce well when printed in black & white and (ii) descriptions of figures in text and captions will be sufficiently clear for both print and online versions. For example, captions should contain the statement "(Color online)." If one desires color in both versions, these considerations are irrelevant, although the authors must guarantee that mandatory additional publication charges will be paid.]

(10) Supplemental files (if any) that might help the reviewers in making their reviews. If the reading of the paper requires prior reading of another paper that has been accepted for publication, but has not yet appeared in print, then PDF file for that manuscript should be included as a supplementary file. Also, if the work draws heavily on previously published material which, while available to the general public, would be time-consuming or possibly expensive for the reviewers to obtain, then PDF files of such relevant material should be included.

(11) Archival supplemental materials to be published with the manuscript in AIP Publishing's Supplemental Materials electronic depository.

In regard to the decision as to what formats one should use for the manuscript and the figures, a principal consideration may be that the likelihood of the published manuscript being more nearly to one's satisfaction is considerably increased if AIP Publishing, during the production process, can make full or partial use of the files you submit. There are conversion programs, for example, that will convert LaTeX and MS Word files to the typesetting system that AIP Publishing uses. If your manuscript is not in either of these formats, then it will be completely retyped. If the figures are submitted in EPS, PS, JPEG, or TIFF format, then they will probably be used directly, at least in part. The uploading of figures in PDF format is not allowed.

D. Steps in online submission

After logging in, one is brought to the Peer X-Press *Task Page* and can select the option of submitting a new manuscript. The resulting process leads the corresponding author through a sequence of screens.

The first screen will display a series of tabs including: Files, Manuscript Information, Confirm Manuscript, and Submit. Clicking on these tabs displays the tasks that must be completed for each step in the submission. Red arrows denote steps that have not been completed. Green arrows are displayed for each tab where the step has been successfully completed.

After submission, all of the individual files, text and tables, plus figures, that make up the full paper will be merged into a single PDF file. One reason for having such a file is that it will generally require less computer memory space. Another is that files in this format are easily read with any computer system. However, the originally submitted set of files, given the acceptance for publication, will be what is submitted to the Production Editing office for final processing.

E. Quality check by editorial office

Upon receiving system notification of a submission, staff members in the Editorial Office check that the overall submission is complete and that the files are properly prepared and suitable for making them available to the Associate Editors and the reviewers. They also check on the estimated length of the manuscript in the event that the author indicates that page charges will not be paid. If all is in order, the Manuscript Coordinator initiates the process, using the ASA-PACS numbers and suggested Associate Editor list supplied by the author, to recruit an Associate Editor who is willing to handle the manuscript. At this time the author also receives a "confirmation of receipt" e-mail message. If the staff members deem that there are submission defects that should be addressed, then the author receives a "quality check" e-mail message. If there are only a small number of defects, the e-mail message may give an explicit description of what is needed. In some cases, when they are very numerous, and it is apparent that the author(s) are not aware that the *Journal* has a set of format requirements, the e-mail message may simply ask the authors to read the instructions (i.e., the present document) and to make a reasonable attempt to follow them.

III. PUBLICATION CHARGES

A. Mandatory charges

Papers of longer length or with color figures desired for the print version of the *Journal* will not be published unless it is first agreed that certain charges will be paid. If it is evident that there is a strong chance that a paper's published length will exceed 12 pages, the paper will not be processed unless the authors guarantee that the charges will be paid. If the paper's published length exceeds 12 pages or more, there is a mandatory charge of $80 per page for the entire article. (The mandatory charge for a 13 page article, for example, would be $1,080, although there would be no mandatory charge if the length were 12 pages.)

To estimate the extent of the page charges, count 3 manuscript pages (double-spaced lines, with wide margins) as equivalent to one printed page, and count 4 figures or tables as equivalent to one printed page. If this number exceeds 12 and your institution and/or sponsor will not pay the page charges, please shorten your paper before submitting it.

Color figures can be included in the online version of the *Journal* with no extra charge, providing that these appear suitably as black and white figures in the print version.

The charges for inclusion of color figures in the print version of the *Journal* are $325 per figure file. If figures that contain parts are submitted in separate files for each part, the $325 charge applies to each file.

If an author's institution or research sponsor is unwilling to pay such charges, the author should make sure that all of the figures in the paper are suitable for black and white printing, and that the estimated length is manifestly such that it will not lead to a printed paper that exceeds 12 pages.

B. Optional charges

To encourage a large circulation of the *Journal* and to allow the inclusion of a large number of selected research articles within its volumes, the *Journal* seeks partial subsidization from the authors and their institutions. Ordinarily, it is the institutions and/or the sponsors of the research that undertake the subsidization. Individual authors must ask their institutions or whatever agencies sponsor their research to pay a page charge of $80 per printed page to help defray the publication costs of the *Journal.* (This is roughly 1/3 of the actual cost per page for the publication of the *Journal.)* The institutions and the sponsoring agencies have the option of declining, although a large fraction of those asked do pay them. The review and selection of manuscripts for publication proceeds without any consideration on the part of the Associate Editors as to whether such page charges will be honored. The publication decision results after consideration of the factors associated with peer review; the acceptance of the page charges is irrelevant.

C. Payment of publication charges—Rightslink

When your page proofs are ready for your review, you will receive an e-mail from AIP Publishing Production Services. It will include a link to an online Rightslink site where you can pay your voluntary or mandatory page charges, color figure charges, or to order reprints of your article. If you are unable to remit payment online, you will find instructions for requesting a printed invoice so that you may pay by check or wire transfer.

IV. FORMAT REQUIREMENTS FOR MANUSCRIPTS

A. Overview

For a manuscript submitted by the online procedure to pass the initial quality control, it is essential that it adhere to a general set of formatting requirements. Such vary from journal to journal, so one should not assume that a manuscript appropriate for another journal's requirements would be satisfactory for the *Journal of the Acoustical Society of America.* The reasons for the *Journal's* requirements are partly to insure a uniform style for publications in the *Journal* and partly to insure that the copy-editing process will be maximally effective in producing a quality publication. For the latter reason, adequate white space throughout the manuscript is desired to allow room for editorial corrections, which will generally be hand-written on a printed hard-copy. While some submitted papers will need very few or no corrections, there is a sufficiently large number of accepted papers of high technical merit that need such editing to make it desirable that all submissions are in a format that amply allows for this.

The following is a list of some of the more important requirements. (More detailed requirements are given in the sections that follow.)

(1) The manuscript must be paginated, starting with the first page.
(2) The entire manuscript must be doubled-spaced. This includes the author addresses, the abstract, the references, and the list of figure captions. It should contain no highlighting.
(3) The title and author list is on the first page. The abstract is ordinarily on a separate page (the second page) unless there is sufficient room on the title page for it, within the constrains of ample margins, 12 pt type, double-spacing, and ample white space. The introduction begins on a separate page following the page that contains the abstract.
(4) The title must be in lower case, with the only capitalized words being the first word and proper nouns.
(5) No acronyms should be in the title or the running title unless they are so common that they can be found in standard dictionaries or unless they are defined in the title.
(6) No unsupported claims for novelty or significance should appear in the title or abstract, such as the use of the words *new, original, novel, important,* and *significant.*
(7) The abstract should be one paragraph and should be limited to 200 words (100 words for Letters to the Editor).
(8) Major section headings should be numbered by capital roman numerals, starting with the introduction. Text of such headings should be in capital letters.
(9) Reference citations should include the full titles and page ranges of all cited papers.
(10) There should be no personal pronouns in the abstract.
(11) No more than one-half of the references should be to the authors themselves.
(12) The total number of figures should not ordinarily be more than 20 (See section V. H).
(13) Line numbers to assist reviewers in commenting on the manuscript may be included but they are not mandatory.

B. Keyboarding instructions

Each submitted paper, even though submitted online, should correspond to a hard copy manuscript. The electronic version has to be prepared so that whatever is printed-out will correspond to the following specifications:

(1) The print-out must be single sided.
(2) The print-out must be configured for standard US letter paper (8.5″ by 11″).
(3) The text on any given page should be confined to an area not to exceed 6.5″ by 9″. (One inch equals 2.54 cm.) All of the margins when printed on standard US letter paper should be at least 1″.
(4) The type font must be 12 pt, and the line spacing must correspond to double spacing (approximately 1/3″ or 0.85 cm per line of print). The fonts used for the text must be of a commonly used easily readable variety such as Times, Helvetica, New York, Courier, Palatino, and Computer Modern.
(5) The authors are requested to use computers with adequate word-processing software in preparing their manuscripts. Ideally, the software must be sufficiently complete that all special symbols used in the manuscript are printed. (The list of symbols available to AIP Publishing for the publication of manuscripts includes virtually all symbols that one can find in modern scientific literature. Authors should refrain from inventing their own symbols.) Italics are similarly designated with a single straight underline in black pencil. It is preferred that vectors be designated by bold face symbols within a published paper rather than by arrows over the symbols.
(6) Manuscript pages must be numbered consecutively, with the title page being page 1.

C. Order of pages

The manuscript pages must appear in the following order:

(1) Title page. (This includes the title, the list of authors, their affiliations, with one complete affiliation for each author appearing immediately after the author's name, an abbreviated title for use as a running title in the published version, and any appropriate footlines to title or authors.)
(2) Abstract page, which may possibly be merged with the title page if there is sufficient room. (This includes the abstract with a separate line giving a prioritized listing of the ASA-PACS numbers that apply to the manuscript. The selected PACS numbers should be taken only from the appendix concerned with acoustics of the overall PACS listing.) Please note that the *Journal* requires the abstract to be typed double spaced, just as for all of the remainder of the manuscript.
(3) Text of the article. This must start on a new page.
(4) Acknowledgments.
(5) Appendixes (if any).
(6) Textual footnotes. (Allowed only if the paper cites references by author name and year of publication.)
(7) References. (If the paper cites references by labeling them with numbers according to the order in which they appear, this section will also include textual footnotes.)
(8) Tables, each on a separate page and each with a caption that is placed above the table.
(9) Collected figure captions.

Figures should ordinarily not be included in the "Article" file. Authors do, however, have the option of including figures embedded in the text, providing there is no ambiguity in distinguishing figure captions from the manuscript text proper. This is understood to be done only for the convenience of the reviewers. Such embedded figures will be ignored in the production editing process. The figures that will be used are those that were uploaded, one by one as separate files, during the online submission process.

D. Title page of manuscript

The title page should include on separate lines, with appropriate intervening spacing: The article title, the name(s) of author(s), one complete affiliation for each author, and the date on which the manuscript is uploaded to the JASA manuscript submission system.

With a distinctive space intervening, the authors must give, on a separate line, a suggested running title of six words or less that contains a maximum of 50 characters. The running title will be printed at the bottom of each printed page, other than the first, when the paper appears in the *Journal.* Because the printing of running titles follows an abbreviated identification of the authors, the maximum permissible length depends critically on the number of the authors and the lengths of their names. The running title also appears on the front cover of the *Journal* as part of an abbreviated table of contents, and it is important that it give a nontrivial indication of the article's content, although some vagueness is to be expected.

Titles should briefly convey the general subject matter of the paper and should not serve as abstracts. The upper limit is set at 17 words. They must be written using only words and terminology that can be found in standard unabridged US English dictionaries or in standard scientific/technical dictionaries, and they must contain no acronyms other than those that can be found in such dictionaries. (If authors believe that the inclusion of a less common acronym in the title will help in information retrieval and/or will help some readers to better understand what is the subject matter of the paper, then that acronym should be explicitly defined in the title.) Ideally, titles should be such that one can easily identify the principal ASA-PACS numbers for the paper, and consequently they should contain appropriate key words. This will enable a reader doing a computer-assisted search to determine whether the paper has any relevance to a given research topic. Begin the first word of the title with a capital letter; thereafter capitalize only proper nouns. The *Journal* does not allow the use of subjective words such as "original," "new," "novel," "important," and "significant" in the title. In general, words whose sole purpose is to tout the importance of a work are regarded as unnecessary; words that clarify the nature of the accomplishment are preferred.

In the list of authors, to simplify later indexing, adopt one form of each name to use on the title pages of all submissions to the *Journal.* It is preferred that the first name be spelled out, especially if the last name is a commonly encountered last name. If an author normally uses the middle name instead of the first name, then an appropriate construction would be one such as J. John Doe. Names must be written with last name (family name) given last. Omit titles such as Professor, Doctor, Colonel, Ph.D., and so on.

Each author may include only one current affiliation in the manuscript. Put the author's name above the institutional affiliation. When there is more than one author with the same institutional affiliation, put all such names above the stating of that affiliation. (See recent issues of the *Journal* for examples.)

In the stating of affiliations, give sufficient (but as briefly as possible) information so that each author may be contacted by mail by interested readers; e-mail addresses are optional. Do not give websites, telephone numbers, or FAX numbers. Names of states and countries should be written out in full. If a post office box should be indicated, append this to the zip code (as in 02537-0339). Use no abbreviations other than D.C. (for District of Columbia). If the address is in the United States, omit the country name.

The preferred order of listing of authors is in accord with the extent of their contributions to the research and to the actual preparation of the manuscript. (Thus, the last listed author is presumed to be the person who has done the least.)

The stated affiliation of any given author should be that of the institution that employed the author at the time the work was done. In the event an author was employed simultaneously by several institutions, the stated affiliation should be that through which the financial support for the research was channeled. If the current (at the time of publication) affiliation is different, then that should be stated in a footline. If an author is deceased then that should be stated in a footline. (Footlines are discussed further below.)

There is no upper limit to the number of authors of any given paper. If the number becomes so large that the appearance of the paper when in print could look excessively awkward, the authors will be given the option of not explicitly printing the author affiliations in the heading of the paper. Instead, these can be handled by use of footlines as described below. The *Journal* does not want organizations or institutions to be listed as authors. If there are a very large number of authors, those who made lesser contributions can be designated by a group name, such a name ending with the word "group." A listing of the members of the group possibly including their addresses should be given in a footline.

Footlines to the title and to the authors' names are consecutively ordered and flagged by lower case alphabetical letters, as in Fletcher[a)], Hunt[b)], and Lindsay[c)]. If there is any history of the work's being presented or published in part earlier, then a footline flag should appear at the end of the title, and the first footline should be of the form exemplified below:[1]

[a)]Portions of this work were presented in "A modal distribution study of violin vibrato," Proceedings of International Computer Music Conference, Thessaloniki, Greece, September 1997, and "Modal distribution analysis of vibrato in musical signals," Proceedings of SPIE International Symposium on Optical Science and Technology, San Diego, CA, July 1998.

Authors have the option of giving a footline stating the e-mail address of one author only (usually the corresponding author), with an appropriate footline flag after that name and with each footline having the form:

[b)]Author to whom correspondence should be addressed. Electronic mail: name@servername.com

E. Abstract page

Abstracts are often published separately from actual articles, and thus are more accessible than the articles themselves to many readers. Authors consequently must write abstracts so that readers without immediate access to the entire article can decide whether the article is worth obtaining. The abstract is customarily written last; the choice of what should be said depends critically on what is said in the body of the paper itself.

The abstract should not be a summary of the paper. Instead, it should give an accurate statement of the subject of the paper, and it should be written so that it is intelligible to a broad category of readers. Explicit results need not be stated, but the nature of the results obtained should be stated. Bear in mind that the abstract of a journal article, unlike the abstract of a talk for a meeting, is backed-up by a written article that is readily (if not immediately) accessible to the reader.

Limit abstracts to 200 words (100 words for Letters to the Editor). Displayed equations that are set apart from the text count as 40 words. Do not use footnotes. If the authors decide that it is imperative to cite a prior publication in the abstract, then the reference should be embedded within the text and enclosed within square brackets. These should be in one of the two standard JASA formats discussed further below, but titles of articles need not be given. The abstract should contain no acknowledgments. In some circumstances, abstracts of longer than 200 words will be allowed. If an author believes that a longer abstract is essential for the paper, they should send an e-mail message to jasa@aip.org with the subject line "Longer abstract requested." The text of the desired abstract should be included in the memo, along with a statement of why the author believes the longer abstract is essential. The abstract will be reviewed by the editors, and possibly a revised wording may be suggested.

Personal pronouns and explicit claims as to novelty should be assiduously avoided. Do not repeat the title in the abstract, and write the abstract with the recognition that the reader has already read the title. Avoid use of acronyms and unfamiliar abbreviations. If the initial writing leads to the multiple use of a single lengthy phrase, avoid using an author-created acronym to achieve a reduction in length of the abstract. Instead, use impersonal pronouns such as *it* and *these* and shorter terms to allude to that phrase. The shortness of the abstract reduces the possibility that the reader will misinterpret the allusion.

On the same page of the abstract, but separated from the abstract by several blank lines, the authors must give the principal ASA-PACS number for the paper, followed by up to three other ASA-PACS numbers that apply. This should be in the format exemplified below:

PACS numbers: 43.30.Pc, 43.30.Sf

The principal ASA-PACS number must be the first in this list. All of the selected PACS numbers must begin with the number 43, this corresponding to the appendix of the overall PACS listing that is concerned with acoustics. Authors are requested not to adopt a principal PACS number in the category of General Linear Acoustics (one beginning with 43.20) unless there is no specific area of acoustics with which the subject matter can be associated. The more specific is the principal PACS number, the greater likelihood that an appropriate match may be made with an Associate Editor, and the greater likelihood that appropriate reviewers will be recruited. When the paper is printed, the list of ASA-PACS numbers will be immediately followed on the same line by the initials, enclosed in brackets, of the Associate Editor who handled the manuscript.

F. Section headings

The text of a manuscript, except for very short Letters to the Editor, is customarily broken up into sections. Four types of section headings are available: principal heading, first subheading, second subheading, and third subheading. The principal headings are typed boldface in all capital letters and appear on separate lines from the text. These are numbered by uppercase roman numerals (I, II, III, IV, etc.), with the introductory section being principal section I. First subheadings are also typed on separate lines; these are numbered by capital letters: A, B, C, etc. The typing of first subheadings is bold-face, with only the first word and proper nouns being capitalized. Second subheadings are ordered by numbers (1, 2, 3, etc.) and are also typed on separate lines. The typing of second subheadings is italic bold-face, also with only the first word and proper nouns capitalized. Third subheadings appear in the text at the beginning of paragraphs. These are numbered by lower case letters (a, b, c, etc.) and these are typed in italics (not bold-faced). Examples of these types of headings can be found in recent issues of the *Journal.* (In earlier issues, the introduction section was not numbered; it is now required to be numbered as the first principal section.)

Headings to appendixes have the same form as principal headings, but are numbered by upper-case letters, with an optional brief title following the identification of the section as an appendix, as exemplified below:

APPENDIX C: CALCULATION OF IMPEDANCES

If there is only one appendix, the letter designation can be omitted.

V. STYLE REQUIREMENTS

A. Citations and footnotes

Regarding the format of citations made within the text, authors have two options: (1) textual footnote style and (2) alphabetical bibliographic list style.

In the *textual footnote style,* references and footnotes are cited in the text by superscripted numerals, as in "the basic equation was first derived by Rayleigh[44] and was subsequently modified by Plesset[45]." References and footnotes to text material are intercalated and numbered consecutively in order of first appearance. If a given reference must be cited at different places in the text, and the citation is identical in all details, then one must use the original number in the second citation.

In the *alphabetical bibliographic list style,* footnotes as such are handled as described above and are intended only to explain or amplify remarks made in the text. Citations to specific papers are flagged by parentheses that enclose either the year of publication or the author's name followed by the year of publication, as in the phrases "some good theories exist (Rayleigh, 1904)" and "a theory was advanced by Rayleigh (1904)." In most of the papers where this style is elected there are no footnotes, and only a bibliographic list ordered alphabetically by the last name of the first author appears at the end of the paper. In a few cases,[2] there is a list of footnotes followed by an alphabetized reference list. Within a footnote, one has the option of referring to any given reference in the same manner as is done in the text proper.

Both styles are in common use in other journals, although the *Journal of the Acoustical Society of America* is one of the few that allows authors a choice. Typically, the textual footnote style is preferred for articles with a smaller number of references, while the alphabetical bibliographic list style is preferred for articles with a large number of references. The diversity of the articles published in the *Journal* makes it infeasible to require just one style unilaterally.

B. General requirements for references

Regardless of what reference style the manuscript uses, the format of the references must include the titles of articles. For articles written in a language other than English, and for which the Latin alphabet is used, give the actual title first in the form in which it appeared in the original reference, followed by the English translation enclosed within parentheses. For titles in other languages, give only the English translation, followed by a statement enclosed in parentheses identifying the language of publication. Do not give Latin-alphabet transliterations of the original title. For titles in English and for English translations of titles, use the same format as specified above for the typing of the title on the title page. Begin the first word of the title with a capital letter; thereafter capitalize only those words that are specified by standard dictionaries to be capitalized in ordinary prose.

One must include only references that can be obtained by the reader. In particular, do not include references that merely state: "personal communication." (Possibly, one can give something analogous to this in a textual footnote, but only as a means of crediting an idea or pinpointing a source. In such a case an explanatory sentence or sentence fragment is preferred to the vague term of "personal communication.") One should also not cite any paper that has only been submitted to a journal; if it has been accepted, then the citation should include an estimated publication date. If one cites a reference, then the listing must contain enough information that the reader can obtain the paper. If thesis, reports, or proceedings are cited, then the listing must contain specific addresses to which one can write to buy or borrow the reference. In general, write the paper in such a manner that its understanding does not depend on the reader having access to references that are not easily obtained.

Authors should avoid giving references to material that is posted on the internet, unless the material is truly archival, as is the case for most online journals. If referring to nonarchival material posted on the internet is necessary to give proper credit for priority, the authors should give the date at which they last viewed the material online. If authors have supplementary material that would be of interest to the readers of the article, then a proper posting of this in an archival form is to make use of the AIP Publishing's supplemental material electronic depository. Instructions for how one posts material can be found at the site <http://scitation.aip.org/content/asa/journal/jasa/info/authors>. Appropriate items for deposit include multimedia (e.g., movie files, audio files, animated .gifs, 3D rendering files), color figures, data tables, and text (e.g., appendices) that are too lengthy or of too limited interest for inclusion in the printed journal. If authors desire to make reference to materials posted by persons other than by the authors, and if the posting is transitory, the authors should first seek to find alternate references of a more archival form that they might cite instead. In all cases, the reading of any material posted at a transitory site must not be a prerequisite to the understanding of the material in the paper itself, and when such material is cited, the authors must take care to point out that the material will not necessarily be obtainable by future readers.

In the event that a reference may be found in several places, as in the print version and the online version of a journal, refer first to the version that is most apt to be archived.

In citing articles, give both the first and last pages that include it. Including the last page will give the reader some indication of the magnitude of the article. The copying en toto of a lengthy article, for example, may be too costly for the reader's current purposes, especially if the chief objective is merely to obtain a better indication of the actual subject matter of the paper than is provided by the title.

The use of the expression *"et al."* in listing authors' names is encouraged in the body of the paper, but must not be used in the actual listing of references, as reference lists in papers are the primary sources of large data bases that persons use, among other purposes, to search by author. This rule applies regardless of the number of authors of the cited paper.

References to unpublished material in the standard format of other references must be avoided. Instead, append a graceful footnote or embed within the text a statement that you are making use of some material that you have acquired from another person—whatever material you actually use of this nature must be peripheral to the development of the principal train of thought of the paper. A critical reader will not accept its validity without at least seeing something in print. If the material is, for example, an unpublished derivation, and if the derivation is important to the substance of the present paper, then repeat the derivation in the manuscript with the original author's permission, possibly including that person as a coauthor.

Journal titles must ordinarily be abbreviated, and each abbreviation must be in a "standard" form. For determination of what abbreviations to use for journals not on the list, one can skim the reference lists that appear at the ends of recent articles in the *Journal.* The general style for making such abbreviations (e.g., Journal is always abbreviated by "J.," Applied is always abbreviated by "Appl.," International is always abbreviated by "Int.," etc.) must in any event emerge from a study of such lists, so the authors should be able to make a good guess as to the standard form. Should the guess be in error, this will often be corrected in the copy-editing process. Egregious errors are often made when the author lifts a citation from another source without actually looking up the original source. An author might be tempted, for example, to abbreviate a journal title as "Pogg. Ann.," taking this from some citation in a 19th century work. The journal cited is *Annalen der Physik,* sometimes published with the title *Annalen der Physik und Chemie,* with the standard abbreviation being "Ann. Phys. (Leipzig)." The fact that J. C. Poggendorff was at one time the editor of this journal gives very little help in the present era in distinguishing it among the astronomical number of journals that have been published. For Poggendorff's contemporaries, however, "Pogg. Ann." had a distinct meaning.

Include in references the names of publishers of book and standards and their locations. References to books and proceedings must include chapter numbers and/or page ranges.

C. Examples of reference formats

The number of possible nuances in the references that one may desire to cite is very large, and the present document cannot address all of them; a study of the reference lists at the ends of articles in recent issues in the *Journal* will resolve most questions. The following two lists, one for each of the styles mentioned above, give some representative examples for the more commonly encountered types of references. If the authors do not find a definitive applicable format in the examples below or in those they see in scanning past issues, then it is suggested that they make their best effort to create an applicable format that is consistent with the examples that they have seen, following the general principles that the information must be sufficiently complete that: (1) any present or future reader can decide whether the work is worth looking at in more detail; (2) such a reader, without great effort, can look at, borrow, photocopy, or buy a copy of the material; and (3) a citation search, based on the title, an author name, a journal name, or a publication category, will result in the present paper being matched with the cited reference.

1. Textual footnote style

[1] Y. Kawai, Prediction of noise propagation from a depressed road by using boundary integral equations" (in Japanese), J. Acoust. Soc. Jpn. **56**, 143–147 (2000).

[2] L. S. Eisenberg, R. V. Shannon, A. S. Martinez, J. Wygonski, and A. Boothroyd, "Speech recognition with reduced spectral cues as a function of age," J. Acoust. Soc. Am. **107**, 2704–2710 (2000).

[3] J. B. Pierrehumbert, *The Phonology and Phonetics of English Intonation* (Ph.D. dissertation, Mass. Inst. Tech., Cambridge, MA, 1980); as cited by 4D. R. Ladd, I. Mennen, and A. Schepman, J. Acoust. Soc. Am. **107**, 2685–2696 (2000).

[4] F. A. McKiel, Jr., "Method and apparatus or sibilant classification in a speech recognition system," U. S. Patent No. 5,897,614 (27 April 1999). A brief review by D. L. Rice appears in: J. Acoust. Soc. Am. 107, p. 2323 (2000).

[5] A. N. Norris, "Finite-amplitude waves in solids, in *Nonlinear Acoustics,* edited by M. F. Hamilton and D. T. Blackstock (Academic Press, San Diego, 1998), Chap. 9, pp. 263–277.

[6] V. V. Muzychenko and S. A. Rybak, "Amplitude of resonance sound scattering by a finite cylindrical shell in a fluid" (in Russian), Akust. Zh. **32**, 129–131 (1986); English transl.: Sov. Phys. Acoust. **32**, 79–80 (1986).

[7] M. Stremel and T. Carolus, "Experimental determination of the fluctuating pressure on a rotating fan blade," on the CD-ROM: *Berlin, March 14–19, Collected Papers, 137th Meeting of the Acoustical Society of America and the 2nd Convention of the European Acoustics Association* (ISBN 3-9804458-5-1, available from Deutsche Gesellschaft fuer Akustik, Fachbereich Physik, Universitaet Oldenburg, 26111 Oldenburg, Germany), paper 1PNSB_7.

[8] ANSI S12.60-2002 (R2009) American National Standard Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools (American National Standards Institute, New York, 2002).

2. Alphabetical bibliographic list style

American National Standards Inst. (2002). ANSI S12.60 (R2009) American National Standard Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools (American National Standards Inst., New York).

Ando, Y. **(1982)**. "Calculation of subjective preference in concert halls," J. Acoust. Soc. Am. Suppl. 1 **71,** S4-S5.

Bacon, S. P. **(2000)**. "Hot topics in psychological and physiological acoustics: Compression," J. Acoust. Soc. Am. **107,** 2864(A).

Bergeijk, W. A. van, Pierce, J. R., and David, E. E., Jr. **(1960).** *Waves and the Ear* (Doubleday, Garden City, NY), Chap. 5, pp. 104-143.

Flatté, S. M., Dashen, R., Munk, W. H., Watson, K. M., and Zachariasen, F. **(1979)**. *Sound Transmission through a Fluctuating Ocean* (Cambridge University Press, London), pp. 31-47.

Hamilton, W. R. **(1837)**. "Third supplement to an essay on the theory of systems of waves," Trans. Roy. Irish Soc. **17** (part 1), 1-144; reprinted in: *The Mathematical Papers of Sir William Rowan Hamilton, Vol. II: Dynamics,* edited by A. W. Conway and A. J. McConnell (Cambridge University Press, London), pp. 162-211.

Helmholtz, H. **(1859)**. "Theorie der Luftschwingungen in Röhren mit offenen Enden" ("Theory of air oscillations in tubes with open ends"), J. reine ang. Math. **57,** 1-72.

Kim, H.-S., Hong, J.-S., Sohn, D.-G., and Oh., J.-E. **(1999)**. "Development of an active muffler system for reducing exhaust noise and flow restriction in a heavy vehicle," Noise Control Eng. J. **47,** 57-63.

Simpson, H. J., and Houston, B. H. **(2000)**. "Synthetic array measurements for waves propagating into a water-saturated sandy bottom ...," J. Acoust. Soc. Am. **107,** 2329-2337.

Other examples may be found in the reference lists of papers recently published in the *Journal.*

D. Figure captions

The illustrations in the *Journal* have *figure captions* rather than *figure titles.* Clarity, rather than brevity, is desired, so captions can extend over several lines. Ideally, a caption must be worded so that a casual reader, on skimming an article, can obtain some indication as to what an illustration is depicting, without actually reading the text of the article. If an illustration is taken from another source, then the caption must acknowledge and cite that source. Various examples of captions can be found in the articles that appear in recent issues of the *Journal.*

If the figure will appear in black and white in the printed edition and in color online, the statement "(Color online)" should be added to the figure caption. For color figures that will appear in black and white in the printed edition of the *Journal,* the reference to colors in the figure may not be included in the caption, e.g., red circles, blue lines.

E. Acknowledgments

The section giving acknowledgments must not be numbered and must appear following the concluding section. It is preferred that acknowledgments be limited to those who helped with the research and with its formulation and to agencies and institutions that provided financial support. Administrators, administrative assistants, associate editors, and persons who assisted in the nontechnical aspects of the manuscript preparation must not be acknowledged. In many cases, sponsoring agencies require that articles give an acknowledgment and specify the format in which the acknowledgment must be stated—doing so is fully acceptable. Generally, the *Journal* expects that the page charges will be honored for any paper that carries an acknowledgment to a sponsoring organization.

F. Mathematical equations

Authors are expected to use computers with appropriate software to typeset mathematical equations.

Authors are also urged to take the nature of the actual layout of the journal pages into account when writing mathematical equations. A line in a column of text is typically 60 characters, but mathematical equations are often longer. To insure that their papers look attractive when printed, authors must seek to write sequences of equations, each of which fits into a single column, some of which define symbols appearing in another equation, even if such results in a greater number of equations. If an equation whose length will exceed that of a single column is unavoidable, then the authors must write the equation so that it is neatly breakable into distinct segments, each of which fits into a single column. The casting of equations in a manner that requires the typesetting to revert to a single column per page (rather than two columns per page) format must be assiduously avoided. To make sure that this possibility will not occur, authors familiar with desk-top publishing software and techniques may find it convenient to temporarily recast manuscripts into a form where the column width corresponds to 60 text characters, so as to see whether none of the line breaks within equations will be awkward.

Equations are numbered consecutively in the text in the order in which they appear, the number designation is in parentheses and on the right side of the page. The numbering of the equations is independent of the section in which they appear for the main body of the text. However, for each appendix, a fresh numbering begins, so that the equations in Appendix B are labeled (B1), (B2), etc. If there is only one appendix, it is treated as if it were Appendix A in the numbering of equations.

G. Phonetic symbols

The phonetic symbols included in a JASA manuscript should be taken from the International Phonetic Alphabet (IPA), which is maintained by the International Phonetic Association, whose home page is http://www.langsci.ucl.ac.uk/ipa/. The display of the most recent version of the alphabet can be found at http://www.langsci.ucl.ac.uk/ipa/ipachart.html.

The total set of phonetic symbols that can be used by AIP Publishing during the typesetting process is the set included among the Unicode characters. This includes most of the symbols and diacritics of the IPA chart, plus a few compiled combinations, additional tonal representations, and separated diacritics. A list of all such symbols is given in the file *phonsymbol.pdf* which can be downloaded by going to the JASA website http://scitation.aip.org/content/asa/journal/jasa/info/authors and then clicking on the item *List of Phonetic Symbols*. This file gives, for each symbol (displayed in 3 different Unicode fonts, DoulosSIL, GentiumPlus, and CharisSILCompact): its Unicode hex ID number, the Unicode character set it is part of, its Unicode character name, and its IPA definition (taken from the IPA chart). Most of these symbols and their Unicode numbers are also available from Professor John Wells of University College London at http://www.phon.ucl.ac.uk/home/wells/ipa-unicode.htm#alfa, without the Unicode character names and character set names.

The method of including such symbols in a manuscript is to use, in conjunction with a word processor, a Unicode-compliant font that includes all symbols required. Fonts that are not Unicode-compliant should not be used. Most computers come with Unicode fonts that give partial coverage of the IPA. Some sources where one can obtain Unicode fonts for Windows, MacOS, and Linux with full IPA coverage are http://www.phon.ucl.ac.uk/home/wells/ipa-unicode.htm and http://scripts.sil.org/cms/scripts/page.php?item_id=SILFontList. Further information about which fonts contain a desired symbol set can be found at http://www.alanwood.net/unicode/fontsbyrange.html#u0250 and adjacent pages at that site. While authors may use any Unicode-compliant font in their manuscript, AIP Publishing reserves the right to replace the author's font with a Unicode font of its choice (currently one of the SIL fonts Doulos, Gentium, or Charis, but subject to change in the future).

For LaTeX manuscripts, PXP's LaTeX-processing environment (MikTeX) supports the use of TIPA fonts. TIPA fonts are available through the Comprehensive TeX Archive Network at http://www.ctan.org/ (download from http://www.ctan.org/pkg/tipa).

H. Figures

Each figure should be manifestly legible when reduced to one column of the printed journal page. Figures requiring the full width of a journal page are discouraged, but exceptions can be made if the reasons for such are sufficiently evident. The inclusion of figures in the manuscript should be such that the manuscript, when published, should ordinarily have no more than 30% of the space devoted to figures, and such that the total number of figures should ordinarily not be more than 20. In terms of the restriction of the total space for figures, each figure part will be considered as occupying a quarter page. Because of the advances in technology and the increasingly wider use of computers in desk-top publishing, it is strongly preferred that authors use computers exclusively in the preparation of illustrations. If any figures are initially in the form of hard copy, they should be scanned with a high quality scanner and converted to electronic form. Each figure that is to be included in the paper should be cast into one of several acceptable formats (TIFF, EPS, JPEG, or PS) and put into a separate file.

The figures are numbered in the order in which they are first referred to in the text. There must be one such referral for every figure in the text. Each figure must have a caption, and the captions are gathered together into a single list that appears at the end of the manuscript. The numbering of the figures, insofar as the online submission process is concerned, is achieved by uploading the individual figure files in the appropriate sequence. The author should take care to make sure that the sequence is correct, but the author will also have the opportunity to view the merged manuscript and to check on this sequencing.

For the most part, figures must be designed so that they will fit within one column (3-3/8″) of the page, and yet be intelligible to the reader. In rare instances, figures requiring full page width are allowed, but the choice for using such a figure must not be capricious.

A chief criticism of many contemporary papers is that they contain far too many computer-generated graphical illustrations that present numerical results. An author develops a certain general computational method (realized by software) and then uses it to exhaustively discuss a large number of special cases. This practice must be avoided. Unless there is an overwhelmingly important single point that the sequence of figures demonstrates as a whole, an applicable rule of thumb is that the maximum number of figures of a given type must be four.

The clarity of most papers is greatly improved if the authors include one or more explanatory sketches. If, for example, the mathematical development presumes a certain geometrical arrangement, then a sketch of this arrangement must be included in the manuscript. If the experiment is carried out with a certain setup of instrumentation and apparatuses, then a sketch is also appropriate. Various clichés, such as Alice's—"and what is the use of a book without pictures?"—are strongly applicable to journal articles in acoustics. The absence of any such figures in a manuscript, even though they might have improved the clarity of the paper, is often construed as an indication of a callous lack of sympathy for the reader's potential difficulties when attempting to understand a paper.

Color figures can be included in the online version of the *Journal* with no extra charge provided that these appear suitably as black and white figures in the print edition.

I. Tables

Tables are numbered by capital roman numerals (TABLE III, TABLE IV, etc.) and are collected at the end of the manuscript, following the references and preceding the figure captions, one table per page. There should be a descriptive caption (not a title) above each table in the manuscript.

Footnotes to individual items in a table are designated by raised lower case letters (0.123[a], Martin[b], etc.) The footnotes as such are given below the table and should be as brief as practicable. If the footnotes are to references already cited in the text, then they should have forms such as— [a]Reference 10—or—[b]Firestone (1935)—depending on the citation style adopted in the text. If the reference is not cited in the text, then the footnote has the same form as a textual footnote when the alphabetical bibliographic list style is used. One would cast the footnote as in the second example above and then include a reference to a 1935 work by Firestone in the paper's overall bibliographic list. If, however, the textual footnote style is used and the reference is not given in the text itself, an explicit reference listing must be given in the table footnote itself. This should contain the bare minimum of information necessary for a reader to retrieve the reference. In general, it is recommended that no footnote refer to references that are not already cited in the text.

VI. THE COVER LETTER

The submission of an electronic file containing a cover letter is now optional. Most of the *Journal's* requirements previously met by the submission of a signed cover letter are now met during the detailed process of online submission. The fact that the manuscript was transmitted by the corresponding author who was duly logged onto the system is taken as prima facie proof that the de facto transmittal letter has been signed by the corresponding author.

There are, however, some circumstances where a cover letter file might be advisable or needed:

(1) If persons who would ordinarily have been included as authors have given permission or requested that their names not be included, then that must be so stated. (This requirement is imposed because some awkward situations have arisen in the past in which persons have complained Information for that colleagues or former colleagues have deliberately omitted their names as authors from papers to which they have contributed. The *Journal* also has the policy that a paper may still be published, even if one of the persons who has contributed to the work refuses to allow his or her name to be included among the list of authors, providing there is no question of plagiarism.) Unless a cover letter listing such exceptions is submitted, the submittal process implies that the corresponding author is attesting that the author list is complete.

(2) If there has been any prior presentation or any overlap in concept with any other manuscripts that have been either published or submitted for publication, this must be stated in a cover letter. If the manuscript has been previously submitted elsewhere for publication, and subsequently withdrawn, this must also be disclosed. If none of these apply for the submitted manuscript, then the submission process is construed to imply that the corresponding author is attesting to such a fact.

(3) (Optional.) Reasons why the authors have selected to submit their paper to JASA rather than some other journal. These would ordinarily be supplied if the authors are concerned that there may be some questions as to the paper

meeting the "truly acoustics" criterion or of its being within the scope of the *Journal.* If none of the references cited in the submitted paper are to articles previously published in the *Journal,* it is highly advisable that some strong reasons be given for why the authors believe the paper falls within the scope of the *Journal.*

(4) If the online submission includes the listing of one or more persons who the authors prefer not be used as reviewers, an explanation in a cover letter would be desirable.

(5) If the authors wish to make statements which they feel are appropriate to be read by editors, but are inappropriate to be included in the actual manuscript, then such should be included in a cover letter.

Cover letters are treated by the Peer X-Press system as being distinct from *rebuttal letters.*

Rebuttal letters should be submitted with revised manuscripts, and the contents are usually such that the authors give, when appropriate, rebuttals to suggestions and criticisms of the reviewers, and give detailed discussion of how and why the revised manuscript differs from what was originally submitted.

VII. EXPLANATIONS AND CATEGORIES

A. Subject classification, ASA-PACS

Authors are asked in their online submittal and in their manuscript to identify the subject classification of their paper using the ASA-PACS system. The subject index of the *Journal* presently follows a specialized extension of the *Physics and Astronomy Classification Scheme*[3] (PACS) maintained by AIP Publishing. Numbers in this scheme pertaining to Acoustics have the general form: 43.nn.Aa, where n denotes a digit, A denotes a capital alphabetical letter, and a denotes a lower case letter. An amplified version of the section 43 listing appears as an appendix to AIP Publishing's document, and this is here referred to as the ASA-PACS system. The ASA-PACS listing for acoustics appears at the end of each volume of the *Journal* preceding the index (June and December issues). It can also be found by first going to the *Journal's* site <http://scitation.aip.org/content/asa/journal/jasa/info/authors> and then clicking the item: *Physics and Astronomy Classification Scheme (PACS), Section 43, Acoustics.* (On the CD distribution of the *Journal,* the appropriate file for the index of each volume is jasin.pdf. The listing of the ASA-PACS numbers is at the beginning of this file.) It is the authors' responsibility to identify a principal ASA-PACS number corresponding to the subject matter of the manuscript and also to identify all other ASA-PACS numbers (up to a total of four) that apply.

B. Suggestions for Associate Editors

In the suggestion of an Associate Editor who should handle a specific manuscript, authors should consult a document titled "Associate Editors identified with PACS classification items" obtainable at the JASA web site <http://scitation.aip.org/content/asa/journal/jasa/info/about>. Here the Associate Editors are identified by their initials, and the relation of the initials to the names is easily discerned from the listing of Associate Editors on the back cover of each issue, on the title page of each volume, and at the online site <http://scitation.aip.org/content/asa/journal/jasa/info/about>. (On the CD distribution of the *Journal,* the appropriate file is jasae.pdf.)

Authors are not constrained to select Associate Editors specifically identified with their choice of principal ASA-PACS number and should note that the *Journal* has special Associate Editors for Mathematical Acoustics, Computational Acoustics, and Education in Acoustics. Review and tutorial articles are ordinarily invited; submission of unsolicited review articles or tutorial articles (other than those which can be construed as papers on education in acoustics) without prior discussion with the Editor-in-Chief is discouraged. Authors should suggest the Associate Editor for Education in Acoustics for tutorial papers that contain material which might be used in standard courses on acoustics or material that supplements standard textbooks.

C. Types of manuscripts

Categories of papers that are published in the *Journal* include the following:

1. Regular research articles

These are papers which report original research. There is neither a lower limit nor an upper limit on their length, although authors must pay page charges if the length results in more than 12 printed pages. The prime requirement is that such papers must contain a complete account of the reported research.

2. Education in acoustics articles

Such papers should be of potential interest to acoustics educators. Examples include descriptions of laboratory experiments and of classroom demonstrations. Papers that describe computer simulations of basic acoustical phenomena also fall within this category. Tutorial discussions on how to present acoustical concepts, including mathematical derivations that might give students additional insight, are possible contributions.

3. Letters to the editor

These are shorter research contributions that can be any of the following: (i) an announcement of a research result, preliminary to the full of the research; (ii) a scientific or technical discussion of a topic that is timely; (iii) brief alternate derivations or alternate experimental evidence concerning acoustical phenomena; (iv) provocative articles that may stimulate further research. Brevity is an essential feature of a letter, and the *Journal* suggests 3 printed journal pages as an upper limit, although it will allow up to 4 printed pages in exceptional cases.

The *Journal's* current format has been chosen so as to give letters greater prominence. Their brevity in conjunction with the possible timeliness of their contents gives impetus to a quicker processing and to a shorter time lag between submission and appearance in printed form in the *Journal.* (The quickest route to publication that the Acoustical Society currently offers is submission to the special section *JASA Express Letters* (JASA-EL) of the *Journal.* For information regarding JASA-EL, visit the site <http://scitation.aip.org/content/asa/journal/jasael/info/authors>.)

Because the desire for brevity is regarded as important, the author is not compelled to make a detailed attempt to place the work within the context of current research; the citations are relatively few and the review of related research is limited. The author should have some reason for desiring a more rapid publication than for a normal article, and the editors and the reviewers should concur with this. The work should have a modicum of completeness, to the extent that the letter "tells a story" that is at least plausible to the reader, and it should have some nontrivial support for what is being related. Not all the loose strings need be tied together. Often there is an implicit promise that the publication of the letter will be followed up by a regular research article that fills in the gaps and that does all the things that a regular research article should do.

4. Errata

These must be corrections to what actually was printed. Authors must explicitly identify the passages or equations in the paper and then state what should replace them. Long essays on why a mistake was made are not desired. A typical line in an errata article would be of the form: *Equation (23) on page 6341 is incorrect. The correct version is ... .* For detailed examples, the authors should look at previously published errata articles in the *Journal.*

5. Comments on published papers

Occasionally, one or more readers, after reading a published paper, will decide to submit a paper giving comments about that paper. The *Journal* welcomes submissions of this type, although they are reviewed to make sure that the comments are reasonable and that they are free of personal slurs. The format of the title of a comments paper is rigidly prescribed, and examples can be found in previous issues of the *Journal.* The authors of the papers under criticism are frequently consulted as reviewers, but their unsubstantiated opinion as to whether the letter is publishable is usually not given much weight.

6. Replies to comments

Authors whose previously published paper has stimulated the submission of a comments paper, and which has subsequently been accepted, have the opportunity to reply to the comments. They are usually (but not invariably) notified of the acceptance of the comments paper, and the *Journal* prefers that the comments and the reply be published in successive pages of the same issue, although this is not always practicable. Replies are also reviewed using criteria similar to those of comments papers. As in the case of comments papers, the format of the title of a reply paper is rigidly prescribed, and examples can be found in the previous issues of the *Journal.*

7. Forum letters

Forum letters are analogous to the "letters to the editor" that one finds in the editorial section of major newspapers. They may express opinions or advocate actions. They may also relate anecdotes or historical facts that may be of general interest to the readers of the *Journal.* They need not have a title and should not have an abstract; they also should be brief, and they should not be of a highly technical nature. These are also submitted using the Peer X-Press system, but are not handled as research articles. The applicable Associate Editor is presently the Editor-in-Chief. For examples of acceptable letters and the format that is desired, prospective authors of such letters should consult examples that have appeared in recent issues of the *Journal.*

8. Tutorial and review papers

Review and tutorial papers are occasionally accepted for publication, but are difficult to handle within the peer-review process. All are handled directly by the Editor-in-Chief, but usually with extensive discussion with the relevant Associate Editors. Usually such are invited, based on recommendations from the Associate Editors and the Technical Committees of the Society, and the tentative acceptance is based on a submitted outline and on the editors' acquaintance with the prospective author's past work. The format of such papers is similar to those of regular research articles, although there should be a table of contents following the abstract for longer research articles. Submission is handled by the online system, but the cover letter should discuss the history of prior discussions with the editors. Because of the large expenditure of time required to write an authorative review article, authors are advised not to begin writing until they have some assurance that there is a good likelihood of the submission eventually being accepted.

9. Book reviews

All book reviews must be first invited by the Associate Editor responsible for book reviews. The format for such reviews is prescribed by the Associate Editor, and the PXP submittal process is used primarily to facilitate the incorporation of the reviews into the Journal.

VIII. FACTORS RELEVANT TO PUBLICATION DECISIONS

A. Peer review system

The *Journal* uses a peer review system in the determination of which submitted manuscripts should be published. The Associate Editors make the actual decisions; each editor has specialized understanding and prior distinguished accomplishments in the subfield of acoustics that encompasses the contributed manuscript. They seek advice from reviewers who are knowledgeable in the general subject of the paper, and the reviewers give opinions on various aspects of the work; primary questions are whether the work is original and whether it is correct. The Associate Editor and the reviewers who examine the manuscript are the authors' peers: persons with comparable standing in the same research field as the authors themselves. (Individuals interested in reviewing for JASA or for JASA-EL can convey that interest via an e-mail message to the Editor-in-Chief at <jasa@aip.org>.)

B. Selection criteria

Many submitted manuscripts are not selected for publication. Selection is based on the following factors: adherence to the stylistic requirements of the *Journal,* clarity and eloquence of exposition, originality of the contribution, demonstrated understanding of previously published literature pertaining to the subject matter, appropriate discussion of the relationships of the reported research to other current research or applications, appropriateness of the subject matter to the *Journal,* correctness of the content of the article, completeness of the reporting of results, the reproducibility of the results, and the significance of the contribution. The *Journal* reserves the right to refuse publication of any submitted article without giving extensively documented reasons, although the editors usually give suggestions that can help the authors in the writing and submission of future papers. The Associate Editor also has the option, but not an obligation, of giving authors an opportunity to submit a revised manuscript addressing specific criticisms raised in the peer review process. The selection process occasionally results in mistakes, but the time limitations of the editors and the reviewers preclude extraordinary steps being taken to insure that no mistakes are ever made. If an author feels that the decision may have been affected by an a priori adverse bias (such as a conflict of interest on the part of one of the reviewers), the system allows authors to express the reasons in writing and ask for an appeal review.

C. Scope of the Journal

Before one decides to submit a paper to the Journal of the Acoustical Society, it is prudent to give some thought as to whether the paper falls within the scope of the Journal. While this can in principal be construed very broadly, it is often the case that another journal would be a more appropriate choice. As a practical matter, the *Journal* would find it difficult to give an adequate peer review to a submitted manuscript that does not fall within the broader areas of expertise of any of its Associate Editors. In the *Journal's* peer-review process, extensive efforts are made to match a submitted manuscript with an Associate Editor knowledgeable in the field, and the Editors have the option of declining to take on the task. It is a tacit understanding that no Associate Editor should accept a paper unless he or she understands the gist of the paper and is able to make a knowledgeable assessment of the relevance of the advice of the selected reviewers. If no one wishes to handle a manuscript, the matter is referred to the Editor-in-Chief and a possible resulting decision is that the manuscript is outside the de facto scope of the *Journal.* When such happens, it is often the case that the article either cites no previously published papers in the *Journal* or else cites no recent papers in any of the other journals that are commonly associated with acoustics. Given that the *Journal* has been in existence for over 80 years and has published of the order of 35,000 papers on a wide variety of acoustical topics over its lifetime, the absence of any references to previously published papers in the *Journal* raises a flag signaling the possibility that the paper lies outside the de facto scope of the *Journal.*

Authors concerned that their work may be construed by the Editors as not being within the scope of the *Journal* can strengthen their case by citing other papers published in the *Journal* that address related topics.

The *Journal* ordinarily selects for publication only articles that have a clear identification with acoustics. It would, for example, not ordinarily publish articles that report results and techniques that are not specifically applicable to acoustics, even though they could be of interest to some persons whose work is concerned with acoustics. An editorial[4] published in the October 1999 issue gives examples that are *not* clearly identifiable with acoustics.

IX. POLICIES REGARDING PRIOR PUBLICATION

The *Journal* adheres assiduously to all applicable copyright laws, and authors must not submit articles whose publication will result in a violation of such laws. Furthermore, the *Journal* follows the tradition of providing an orderly archive of scientific research in which authors take care that results and ideas are fully attributed to their originators. Conscious plagiarism is a serious breach of ethics, if not illegal. (Submission of an article that is plagiarized, in part or in full, may have serious repercussions on the future careers of the authors.) Occasionally, authors rediscover older results and submit papers reporting these results as though they were new. The desire to safeguard the *Journal* from publishing any such paper requires that submitted articles have a sufficient discussion of prior related literature to demonstrate the authors' familiarity with the literature and to establish the credibility of the assertion that the authors have carried out a thorough literature search.

In many cases, the authors themselves may have either previously circulated, published, or presented work that has substantial similarities with what is contained within the contributed manuscript. In general, JASA will not publish work that has been previously published. (An exception is when the previous publication is a letter to the editor, and when pertinent details were omitted because of the brief nature of the earlier reporting.) Presentations at conferences are not construed as prior publication; neither is the circulation of preprints or the posting of preprints on any web site, providing the site does not have the semblance of an archival online journal. Publication as such implies that the work is currently, and for the indefinite future, available, either for purchase or on loan, to a broad segment of the research community. Often the *Journal* will consider publishing manuscripts with tangible similarities to other work previously published by the authors—providing the following conditions are met: (1) the titles are different; (2) the submitted manuscript contains no extensive passages of text or figures that are the same as in the previous publication; (3) the present manuscript is a substantial update of the previous publication; (4) the previous publication has substantially less availability than would a publication in JASA; (5) the current manuscript gives ample referencing to the prior publication and explains how the current manuscript differs from the prior publication. Decisions regarding such cases are made by the Associate Editors, often in consultation with the Editor-in-Chief. (Inquiries prior to submission as to whether a given manuscript with some prior history of publication may be regarded as suitable for JASA should be addressed to the Editor-in-Chief at <jasa@aip.org>.)

The *Journal* will not consider any manuscript for publication that is presently under consideration by another journal or which is substantially similar to another one under consideration. If it should learn that such is the case, the paper will be rejected and the editors of the other journal will be notified.

Authors of an article previously published as a letter to the editor, either as a regular letter or as a letter in the JASA-EL (*JASA Express Letters*) section of the *Journal,* where the original account was either abbreviated or preliminary are encouraged to submit a more comprehensive and up-dated account of their research to the *Journal.*

A. Speculative papers

In some cases, a paper may be largely speculative; a new theory may be offered for an as yet imperfectly understood phenomenon, without complete confirmation by experiment. Although such papers may be controversial, they often become the most important papers in the long-term development of a scientific field. They also play an important role in the stimulation of good research. Such papers are intrinsically publishable in JASA, although explicit guidelines for their selection are difficult to formulate. Of major importance are (i) that the logical development be as complete as practicable, (ii) that the principal ideas be plausible and consistent with what is currently known, (iii) that there be no known counter-examples, and (iv) that the authors give some hints as to how the ideas might be checked by future experiments or numerical computations. In addition, the authors should cite whatever prior literature exists that might indicate that others have made similar speculations.

B. Multiple submissions

The current online submittal process requires that each paper be submitted independently. Each received manuscript will be separately reviewed and judged regarding its merits for publication independently of the others. There is no formal mechanism for an author to request that two submissions, closely spaced in their times of submission, be regarded as a single submission.

In particular, the submission of two manuscripts, one labeled “Part I” and the other labeled “Part II” is not allowed. Submission of a single manuscript with the label “Part I” is also not allowed. An author may submit a separate manuscript labeled “Part II,” if the text identifies which previously accepted paper is to be regarded as “Part I.” Doing so may be a convenient method for alerting potential readers to the fact that the paper is a sequel to a previous paper by the author. The author should not submit a paper so labeled, however, unless the paper to be designated as “Part I” has already been accepted, either for JASA or another journal.

The Associate Editors are instructed not to process any manuscript that cannot be read without the help of as yet unpublished papers that are still under review. Consequently, authors are requested to hold back the submission of “sequels” to previously submitted papers until the disposition of those papers is determined. Alternately, authors should write the “sequels” so that the reading and comprehension of those manuscripts does not require prior reading and access of papers whose publication is still uncertain.

X. SUGGESTIONS REGARDING CONTENT

A. Introductory section

Every paper begins with introductory paragraphs. Except for short Letters to the Editor, these paragraphs appear within a separate principal section, usually with the heading “Introduction.”

Although some discussion of the background of the work may be advisable, a statement of the precise subject of the work must appear within the first two paragraphs. The reader need not fully understand the subject the first time it is stated; subsequent sentences and paragraphs should clarify the statement and should supply further necessary background. The extent of the clarification must be such that a nonspecialist will be able to obtain a reasonable idea of what the paper is about. The introduction should also explain to the nonspecialist just how the present work fits into the context of other current work done by persons other than the authors themselves. Beyond meeting these obligations, the writing should be as concise as practicable.

The introduction must give the authors’ best arguments as to why the work is original and significant. This is customarily done via a knowledgeable discussion of current and prior literature. The authors should envision typical readers or typical reviewers, and this should be a set of people that is not inordinately small, and the authors must write so as to convince them. In some cases, both originality and significance will be immediately evident to all such persons, and the arguments can be brief. In other cases, the authors may have a daunting task. It must not be assumed that readers and reviewers will give the authors the benefit of the doubt.

B. Main body of text

The writing in the main body of the paper must follow a consistent logical order. It should contain only material that pertains to the main premise of the paper, and that premise should have been stated in the introduction. While tutorial discussions may in some places be appropriate, such should be kept to a minimum and should be only to the extent necessary to keep the envisioned readers from becoming lost.

The writing throughout the text, including the introduction, must be in the present tense. It may be tempting to refer to subsequent sections and passages in the manuscript in the future tense, but the authors must assiduously avoid doing so, using instead phrases such as "is discussed further below."

Whenever pertinent results, primary or secondary, are reached in the progress of the paper, the writing should point out that these are pertinent results in such a manner that it would get the attention of a reader who is rapidly scanning the paper.

The requirement of a consistent logical order implies that the logical steps appear in consecutive order. Readers must not be referred to subsequent passages or to appendixes to fill in key elements of the logical development. The fact that any one such key element is lengthy or awkward is insufficient reason to relegate it to an appendix. Authors can, however, flag such passages giving the casual reader the option of skipping over them on first reading. The writing nevertheless must be directed toward the critical reader—a person who accepts no aspect of the paper on faith. (If the paper has some elements that are primarily speculative, then that should be explicitly stated, and the development should be directed toward establishing the plausibility of the speculation for the critical reader.)

To achieve clarity and readability, the authors must explicitly state the purposes of lengthy descriptions or of lengthy derivations at the beginning of the relevant passages. There should be no mysteries throughout the manuscript as to the direction in which the presentation is going.

Authors must take care that no reader becomes needlessly lost because of the use of lesser-known terminology. All terms not in standard dictionaries must be defined when they are first used. Acronyms should be avoided, but, when they are necessary, they must be explicitly defined when first used. The terminology must be consistent; different words should not be used to represent the same concept.

Efforts must be taken to avoid insulting the reader with the use of gratuitous terms or phrases such as *obvious, well-known, evident,* or *trivial.* If the adjectives are applicable, then they are unnecessary. If not, then the authors risk incurring the ill-will of the readers.

If it becomes necessary to bring in externally obtained results, then the reader must be apprised, preferably by an explicit citation to accessible literature, of the source of such results. There must be no vague allusions, such as "It has been found that..." or "It can be shown that..." If the allusion is to a mathematical derivation that the authors have themselves carried out, but which they feel is not worth describing in detail, then they should briefly outline how the derivation can be carried out, with the implication that a competent reader can fill in the necessary steps without difficulty.

For an archival journal such as JASA, reproducibility of reported results is of prime importance. Consequently, authors must give a sufficiently detailed account, so that all results, other than anecdotal, can be checked by a competent reader with comparable research facilities. If the results are numerical, then the authors must give estimates of the probable errors and state how they arrived at such estimates. (Anecdotal results are typically results of field experiments or unique case studies; such are often worth publishing as they can stimulate further work and can be used in conjunction with other results to piece together a coherent understanding of broader classes of phenomena.)

C. Concluding section

The last principal section of the article is customarily labeled "Conclusions" or "Concluding Remarks." This should not repeat the abstract, and it should not restate the subject of the paper. The wording should be directed toward a person who has some, if not thorough, familiarity with the main body of the text and who knows what the paper is all about. The authors should review the principal results of the paper and should point out just where these emerged in the body of the text. There should be a frank discussion of the limitations, if any, of the results, and there should be a broad discussion of possible implications of these results.

Often the concluding section gracefully ends with speculations on what research might be done in the future to build upon the results of the present paper. Here the authors must write in a collegial tone. There should be no remarks stating what the authors themselves intend to do next. They must be careful not to imply that the future work in the subject matter of the paper is the exclusive domain of the authors, and there should be no allusions to work in progress or to work whose publication is uncertain. It is conceivable that readers stimulated to do work along the lines suggested by the paper will contact the authors directly to avoid a duplication of effort, but that will be their choice. The spirit expressed in the paper itself should be that anyone should be free to follow-up on the suggestions made in the concluding section. A successful paper is one that does incite such interest on the part of the readers and one which is extensively cited in future papers written by persons other than the authors themselves.

D. Appendixes

The *Journal* prefers that articles not include appendixes unless there are strong reasons for their being included. Details of mathematical developments or of experimental procedures that are critical to the understanding of the substance of a paper must not be relegated to an appendix. (Authors must bear in mind that readers can easily skim over difficult passages in their first reading of a paper.) Lengthy proofs of theorems may possibly be placed in appendixes providing their stating as such in the main body of the text is manifestly plausible. Short appendixes are generally unnecessary and impede the comprehension of the paper. Appendixes may be used for lengthy tabulations of data, of explicit formulas for special cases, and of numerical results. Editors and reviewers, however, may question whether their inclusion is necessary.

E. Selection of references

References are typically cited extensively in the introduction, and the selection of such references can play an important role in the potential usefulness of the paper to future readers and in the opinions that readers and reviewers form of the paper. No hard and fast rules can be set down as to how authors can best select references and as to how they should discuss them, but some suggestions can be found in an editorial[5] published in the May 2000 issue. If a paper falls within the scope of the *Journal,* one would ordinarily expect to find several references to papers previously published in JASA.

Demonstration of the relevance of the work is often accomplished via citations, with accompanying discussion, to recent articles in JASA and analogous journals. The implied claims to originality can be strengthened via citations, with accompanying discussion, to prior work related to the subject of the paper, sufficient to establish credibility that the authors are familiar with the literature and are not duplicating previous published work. Unsupported assertions that the authors are familiar with all applicable literature and that they have carried out an exhaustive literature survey are generally unconvincing to the critical reader.

Authors must not make large block citations of many references (e.g., four or more). There must be a stated reason for the citation of each reference, although the same reason can sometimes apply simultaneously to a small number of references. The total number of references should be kept as small a number as is consistent with the principal purposes of the paper (45 references is a suggested upper limit for a regular research article). Although nonspecialist readers may find a given paper to be informative in regard to the general state of a given field, the authors must not consciously write a research paper so that it will fulfill a dual function of being a review paper or of being a tutorial paper.

Less literate readers often form and propagate erroneous opinions concerning priority of ideas and discoveries based on the reading of recent papers, so authors must make a conscious attempt to cite original sources. Secondary sources can also be cited, if they are identified as such and especially if they are more accessible or if they provide more readable accounts. In such cases, reasons must be given as to why the secondary sources are being cited. References to individual textbooks for results that can be found in a large number of analogous textbooks should not be given, unless the cited textbook gives a uniquely clear or detailed discussion of the result. Authors should assume that any reader has access to some such textbook, and the authors should tacitly treat the result as well-known and not requiring a reference citation.

Authors must not cite any reference that the authors have not explicitly seen, unless the paper has a statement to that effect, accompanied by a statement of how the authors became aware of the reference. Such citations should be limited to crediting priority, and there must be no implied recommendations that readers should read literature which the authors themselves have not read.

XI. SUGGESTIONS REGARDING STYLE

A. Quality of writing and word usage

The *Journal* publishes articles in the English language only. There are very few differences of substance between British English style (as codified in the *Oxford English Dictionary*[6]) and US English style, but authors frequently must make choices in this respect, such as between alternate spelling of words that end in either *-or* or *-our,* or in either *-ized* or *-ised,* or in either *-er* or *-re.* Although now a de facto international journal, JASA because of its historical origins requires manuscripts to follow US English style conventions.

Articles published in JASA are expected to adhere to high standards of scholarly writing. A formal writing style free of slang is required. Good conversational skills do not necessarily translate to good formal writing skills. Authors are expected to make whatever use is necessary of standard authoritative references in regard to English grammar and writing style in preparing their manuscripts. Many good references exist—among those frequently used by professional writers are Webster's Third New International Dictionary, Unabridged,[7] Merriam-Webster's Collegiate Dictionary, 11th Edition,[8] Strunk and White's Elements of Style,[9] and the Chicago Manual of Style.[10] (The Third New International is AIP Publishing's standard dictionary.) All authors are urged to do their best to produce a high quality readable manuscript, consistent with the best traditions of scholarly and erudite writing. Occasional typographical errors and lapses of grammar can be taken care of in the copy-editing phase of the production process, and the instructions given here are intended that there be ample white space in the printed-out manuscript that such copy-editing can be carried out. Receipt of a paper whose grammatical and style errors are so excessive that they cannot be easily fixed by copy-editing will generally result in the authors being notified that the submission is not acceptable. Receipt of such a notification should not be construed as a rejection of the manuscript—the authors should take steps, possibly with external help, to revise the manuscript so that it overcomes these deficiencies. (Authors needing help or advice on scientific writing in the English language are encouraged to contact colleagues, both within and outside their own institutions, to crititque the writing in their manuscripts. Unfortunately, the staff of the *Journal* does not have the time to do this on a routine basis.)

There are some minor discrepancies in the stylistic rules that are prescribed in various references—these generally arise because of the differences in priorities that are set in different publication categories. Newspapers, for example, put high emphasis on the efficient use of limited space for conveying the news and for catching the interest of their readers. For scholarly journals, on the other hand, the overwhelming priority is clarity. In the references cited above, this is the basis for most of the stated rules. In following this tradition, the *Journal,* for example, requires a rigorous adherence to the serial comma rule (Strunk's rule number 2): In a series of three or more terms with a single conjunction, use a comma after each term except the last. Thus a JASA manuscript would refer to the "theory of Rayleigh, Helmholtz, and Kirchhoff" rather than to the "theory of Rayleigh, Helmholtz and Kirchhoff."

The priority of clarity requires that authors only use words that are likely to be understood by a large majority of potential readers. Usable words are those whose definitions may be found either in a standard unabridged English dictionary (such as the Webster's Third New International mentioned above), in a standard scientific dictionary such as the Academic Press Dictionary of Science and Technology,[11] or in a dictionary specifically devoted to acoustics such as the Dictionary of Acoustics[12] by C. L. Morfey. In some cases, words and phrases that are not in any dictionary may be *in vogue* among some workers in a given field, especially among the authors and their colleagues. Authors must give careful consideration to whether use of such terms in their manuscript is necessary; and if the authors decide to use them, precise definitions must be stated within the manuscript. Unilateral coinage of new terms by the authors is discouraged. In some cases, words with different meanings and with different spellings are pronounced exactly the same, and authors must be careful to choose the right spelling. Common errors are to interchange principal and principle and to interchange role and roll.

B. Grammatical pitfalls

There are only a relatively small number of categories of errors that authors frequently make in the preparation of manuscripts. Authors should be aware of these common pitfalls and double-check that their manuscripts contain no errors in these categories. Some errors will be evident when the manuscript is read aloud; others, depending on the background of the writers, may not be. Common categories are (1) dangling participles, (2) lack of agreement in number (plural versus singular) of verbs with their subjects, (3) omission of necessary articles (such as a, an, and the) that precede nouns, (4) the use of incorrect case forms (subjective, objective, possessive) for pronouns (e.g., who versus whom), and (5) use of the incorrect form (present, past, past participle, and future) in regard to tense for a verb. Individual authors may have their own peculiar pitfalls, and an independent casual reading of the manuscript by another person will generally pinpoint such pitfalls. Given the recognition that such exist, a diligent author should be able to go through the manuscript and find all instances where errors of the identified types occur.

C. Active voice and personal pronouns

Many authorities on good writing emphasize that authors should use the active rather than the passive voice. Doing so in scholarly writing, especially when mathematical expressions are present, is often infeasible, but the advice has merit. In mathematical derivations, for example, some authors use the tutorial we to avoid using the passive voice, so that one writes: "We substitute the expression on the right side of Eq. (5) into Eq. (2) and obtain ...," rather than: "The right side of Eq. (5) is substituted into Eq. (2), with the result being ... ." A preferable construction is to avoid the use of the tutorial we and to use transitive verbs such as yields, generates, produces, and leads to. Thus one would write the example above as: "Substitution of Eq. (5) into Eq. (2) yields ... ." Good writers frequently go over an early draft of a manuscript, examine each sentence and phrase written using the passive voice, and consider whether they can improve the sentence by rewriting it.

In general, personal pronouns, including the "tutorial we," are preferably avoided in scholarly writing, so that the tone is impersonal and dispassionate. In a few cases, it is appropriate that an opinion be given or that a unique personal experience be related, and personal pronouns are unavoidable. What should be assiduously avoided are any egotistical statements using personal pronouns. If a personal opinion needs to be expressed, a preferred construction is to refer to the author in the third person, such as: "the present writer believes that ... ."

D. Acronyms

Acronyms have the inconvenient feature that, should the reader be unfamiliar with them, the reader is clueless as to their meaning. Articles in scholarly journals should ideally be intelligible to many generations of future readers, and formerly common acronyms such as RCA (Radio Corporation of America, recently merged into the General Electric Corporation) and REA (Rural Electrification Authority) may have no meaning to such readers. Consequently, authors are requested to use acronyms sparingly and generally only when not using them would result in exceedingly awkward prose. Acronyms, such as SONAR and LASER (currently written in lower case, sonar and laser, as ordinary words), that have become standard terms in the English language and that can be readily found in abridged dictionaries, are exceptions. If the authors use acronyms not in this category, then the meaning of the individual letters should be spelled out at the time such an acronym is first introduced. An article containing, say, three or more acronyms in every paragraph will be regarded as pretentious and deliberately opaque.

E. Computer programs

In some cases the archival reporting of research suggests that authors give the names of specific computer programs used in the research. If the computation or data processing could just as well have been carried out with the aid of any one of a variety of such programs, then the name should be omitted. If the program has unique features that are used in the current research, then the stating of the program name must be accompanied by a brief explanation of the principal premises and functions on which the relevant features are based. One overriding consideration is that the *Journal* wishes to avoid implied endorsements of any commercial product.

F. Code words

Large research projects and large experiments that involve several research groups are frequently referred to by code words. Research articles in the *Journal* must be intelligible to a much broader group of readers, both present and future, than those individuals involved in the projects with which such a code word is associated. If possible, such code words should either not be used or else referred to in only a parenthetical sense. If attempting to do this leads to exceptionally awkward writing, then the authors must take special care to explicitly explain the nature of the project early in the paper. They must avoid any impression that the paper is specifically directed toward members of some in-group.

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[5]A. D. Pierce, Literate writing and collegial citing, J. Acoust. Soc. Am. **107,** 2303–2311 (2000).

[6]*The Oxford English Dictionary,* edited by J. Simpson and E. Weiner (Oxford University Press, 1989, 2nd edition), 20 volumes. Also published as *Oxford English Dictionary (Second Edition) on CD-ROM, version 2.0* (Oxford University Press, 1999). An online version is available by subscription at the Internet site <http://www.oed.com/public/welcome>.

[7]*Webster's Third New International Dictionary of the English Language, Unabridged,* Philip Babcock Gove, Editor-in-Chief (Merriam-Webster Inc., Springfield, MA, 1993, principal copyright 1961) This is the eighth in a series of dictionaries that has its beginning in Noah Webster's *American Dictionary of the English Language* (1828).

[8]*Merriam-Webster's Collegiate Dictionary, 11th Edition* (Merriam-Webster, Springfield, MA, 2003, principal copyright 1993). (A freshly updated version is issued annually.)

[9]W. Strunk, Jr. and E. B. White, *The Elements of Style,* with forward by Roger Angell (Allyn and Bacon, 1999, 4th edition).

[10]*The Chicago Manual of Style: The Essential Guide for Writers, Editors, and Publishers,* with preface by John Grossman (University of Chicago Press, 1993, 14th edition).

[11]*Academic Press Dictionary of Science and Technology,* edited by Christopher Morris (Academic Press, Inc., 1992).

[12]C. L. Morfey, *Dictionary of Acoustics* (Academic Press, Inc., 2000).

ETHICAL PRINCIPLES OF THE ACOUSTICAL SOCIETY OF AMERICA FOR RESEARCH INVOLVING HUMAN AND NON-HUMAN ANIMALS IN RESEARCH AND PUBLISHING AND PRESENTATIONS

The Acoustical Society of America (ASA) has endorsed the following ethical principles associated with the use of human and non-human animals in research, and for publishing and presentations. The principles endorsed by the Society follow the form of those adopted by the American Psychological Association (APA), along with excerpts borrowed from the Council for International Organizations of Medical Sciences (CIOMS). The ASA acknowledges the difficulty in making ethical judgments, but the ASA wishes to set minimum socially accepted ethical standards for publishing in its journals and presenting at its meetings. These Ethical Principles are based on the principle that the individual author or presenter bears the responsibility for the ethical conduct of their research and is publication or presentation.

Authors of manuscripts submitted for publication in a journal of the Acoustical Society of America or presenting a paper at a meeting of the Society are obligated to follow the ethical principles of the Society. Failure to accept the ethical principles of the ASA shall result in the immediate rejection of manuscripts and/or proposals for publication or presentation. False indications of having followed the Ethical Principles of the ASA may be brought to the Ethics and Grievances Committee of the ASA.

APPROVAL BY APPROPRIATE GOVERNING AUTHORITY

The ASA requires all authors to abide by the principles of ethical research as a prerequisite for participation in Society-wide activities (e.g., publication of papers, presentations at meetings, etc.). Furthermore, the Society endorses the view that all research involving human and non-human vertebrate animals requires approval by the appropriate governing authority (e.g., institutional review board [IRB], or institutional animal care and use committee [IACUC], Health Insurance Portability and Accountability Act [HIPAA], or by other governing authorities used in many countries) and adopts the requirement that all research must be conducted in accordance with an approved research protocol as a precondition for participation in ASA programs. If no such governing authority exists, then the intent of the ASA Ethical Principles described in this document must be met. All research involving the use of human or non-human animals must have met the ASA Ethical Principles prior to the materials being submitted to the ASA for publication or presentation.

USE OF HUMAN SUBJECTS IN RESEARCH-Applicable when human subjects are used in the research

Research involving the use of human subjects should have been approved by an existing appropriate governing authority (e.g., an institutional review board [IRB]) whose policies are consistent with the Ethical Principles of the ASA **or** the research should have met the following criteria:

Informed Consent

When obtaining informed consent from prospective participants in a research protocol that has been approved by the appropriate and responsible-governing body, authors must have clearly and simply specified to the participants beforehand:

1. The purpose of the research, the expected duration of the study, and all procedures that were to be used.
2. The right of participants to decline to participate and to withdraw from the research in question after participation began.
3. The foreseeable consequences of declining or withdrawing from a study.
4. Anticipated factors that may have influenced a prospective participant's willingness to participate in a research project, such as potential risks, discomfort, or adverse effects.
5. All prospective research benefits.
6. The limits of confidentially.
7. Incentives for participation.
8. Whom to contact for questions about the research and the rights of research participants. The office/person must have willingly provided an atmosphere in which prospective participants were able to ask questions and receive answers.

Authors conducting intervention research involving the use of experimental treatments must have clarified, for each prospective participant, the following issues at the outset of the research:

1. The experimental nature of the treatment;
2. The services that were or were not to be available to the control group(s) if appropriate;
3. The means by which assignment to treatment and control groups were made;
4. Available treatment alternatives if an individual did not wish to participate in the research or wished to withdraw once a study had begun; and
5. Compensation for expenses incurred as a result of participating in a study including, if appropriate, whether reimbursement from the participant or a third-party payer was sought.

Informed Consent for Recording Voices and Images in Research

Authors must have obtained informed consent from research participants prior to recording their voices or images for data collection unless:

1. The research consisted solely of naturalistic observations in public places, and it was not anticipated that the recording would be used in a manner that could have caused personal identification or harm, or
2. The research design included deception. If deceptive tactics were a necessary component of the research design, consent for the use of recordings was obtained during the debriefing session.

Client/Patient, Student, and Subordinate Research Participants

When authors conduct research with clients/patients, students, or subordinates as participants, they must have taken steps to protect the prospective participants from adverse consequences of declining or withdrawing from participation.

Dispensing With Informed Consent for Research

Authors may have dispensed with the requirement to obtain informed consent when:

1. It was reasonable to assume that the research protocol in question did not create distress or harm to the participant and involves:
   a. The study of normal educational practices, curricula, or classroom management methods that were conducted in educational settings
   b. Anonymous questionnaires, naturalistic observations, or archival research for which disclosure of responses would not place participants at risk of criminal or civil liability or damage their financial standing, employability, or reputation, and confidentiality
   c. The study of factors related to job or organization effectiveness conducted in organizational settings for which there was no risk to participants' employability, and confidentiality.
2. Dispensation is permitted by law.
3. The research involved the collection or study of existing data, documents, records, pathological specimens, or diagnostic specimens, if these sources are publicly available or if the information is recorded by the investigator in such a manner that subjects cannot be identified, directly or through identifiers linked to the subjects.

Offering Inducements for Research Participation

(a) Authors must not have made excessive or inappropriate financial or other inducements for research participation when such inducements are likely to coerce participation.

(b) When offering professional services as an inducement for research participation, authors must have clarified the nature of the services, as well as the risks, obligations, and limitations.

Deception in Research

(a) Authors must not have conducted a study involving deception unless they had determined that the use of deceptive techniques was justified by the study's significant prospective scientific, educational, or applied value and that effective non-deceptive alternative procedures were not feasible.

(b) Authors must not have deceived prospective participants about research that is reasonably expected to cause physical pain or severe emotional distress.

(c) Authors must have explained any deception that was an integral feature of the design and conduct of an experiment to participants as early as was feasible, preferably at the conclusion of their participation, but no later than at the conclusion of the data collection period, and participants were freely permitted to withdraw their data.

Debriefing

(a) Authors must have provided a prompt opportunity for participants to obtain appropriate information about the nature, results, and conclusions of the research project for which they were a part, and they must have taken reasonable steps to correct any misconceptions that participants may have had of which the experimenters were aware.

(b) If scientific or humane values justified delaying or withholding relevant information, authors must have taken reasonable measures to reduce the risk of harm.

(c) If authors were aware that research procedures had harmed a participant, they must have taken reasonable steps to have minimized the harm.

HUMANE CARE AND USE OF NON-HUMAN VERTEBRATE ANIMALS IN RESEARCH-Applicable when non-human vertebrate animals are used in the research

The advancement of science and the development of improved means to protect the health and well being both of human and non-human vertebrate animals often require the use of intact individuals representing a wide variety of species in experiments designed to address reasonable scientific questions. Vertebrate animal experiments should have been undertaken only after due consideration of the relevance for health, conservation, and the advancement of scientific knowledge. (Modified from the Council for International Organizations of Medical Science (CIOMS) document: "International Guiding Principles for Biomedical Research Involving Animals1985"). Research involving the use of vertebrate animals should have been approved by an existing appropriate governing authority (e.g., an institutional animal care and use committee [IACUC]) whose policies are consistent with the Ethical Principles of the ASA **or** the research should have met the following criteria:

The proper and humane treatment of vertebrate animals in research demands that investigators:

1. Acquired, cared for, used, interacted with, observed, and disposed of animals in compliance with all current federal, state, and local laws and regulations, and with professional standards.
2. Are knowledgeable of applicable research methods and are experienced in the care of laboratory animals, supervised all procedures involving animals, and assumed responsibility for the comfort, health, and humane treatment of experimental animals under all circumstances.
3. Have insured that the current research is not repetitive of previously published work.
4. Should have used alternatives (e.g., mathematical models, computer simulations, etc.) when possible and reasonable.
5. Must have performed surgical procedures that were under appropriate anesthesia and followed techniques that avoided infection and minimized pain during and after surgery.
6. Have ensured that all subordinates who use animals as a part of their employment or education received instruction in research methods and in the care, maintenance, and handling of the species that were used, commensurate with the nature of their role as a member of the research team.
7. Must have made all reasonable efforts to minimize the number of vertebrate animals used, the discomfort, the illness, and the pain of all animal subjects.
8. Must have made all reasonable efforts to minimize any harm to the environment necessary for the safety and well being of animals that were observed or may have been affective as part of a research study.
9. Must have made all reasonable efforts to have monitored and then mitigated any possible adverse affects to animals that were observed as a function of the experimental protocol.
10. Who have used a procedure subjecting animals to pain, stress, or privation may have done so only when an alternative procedure was unavailable; the goal was justified by its prospective scientific, educational, or applied value; and the protocol had been approved by an appropriate review board.
11. Proceeded rapidly to humanely terminate an animal's life when it was necessary and appropriate, always minimizing pain and always in accordance with accepted procedures as determined by an appropriate review board.

PUBLICATION and PRESENTATION ETHICS-For publications in ASA journals and presentations at ASA sponsored meetings

Plagiarism

Authors must not have presented portions of another's work or data as their own under any circumstances.

Publication Credit

Authors have taken responsibility and credit, including authorship credit, only for work they have actually performed or to which they have substantially contributed. Principal authorship and other publication credits accurately reflect the relative scientific or professional contributions of the individuals involved, regardless of their relative status. Mere possession of an institutional position, such as a department chair, does not justify authorship credit. Minor contributions to the research or to the writing of the paper should have been acknowledged appropriately, such as in footnotes or in an introductory statement.

Duplicate Publication of Data

Authors did not publish, as original data, findings that have been previously published. This does not preclude the republication of data when they are accompanied by proper acknowledgment as defined by the publication policies of the ASA.

Reporting Research Results

If authors discover significant errors in published data, reasonable steps must be made in as timely a manner as possible to rectify such errors. Errors can be rectified by a correction, retraction, erratum, or other appropriate publication means.

DISCLOSURE OF CONFLICTS OF INTEREST

If the publication or presentation of the work could directly benefit the author(s), especially financially, then the author(s) must disclose the nature of the conflict:

1) The complete affiliation(s) of each author and sources of funding for the published or presented research should be clearly described in the paper or publication abstract.

2) If the publication or presentation of the research would directly lead to the financial gain of the authors(s), then a statement to this effect must appear in the acknowledgment section of the paper or presentation abstract or in a footnote of a paper.

3) If the research that is to be published or presented is in a controversial area and the publication or presentation presents only one view in regard to the controversy, then the existence of the controversy and this view must be provided in the acknowledgment section of the paper or presentation abstract or in a footnote of a paper. It is the responsibility of the author to determine if the paper or presentation is in a controversial area and if the person is expressing a singular view regarding the controversy.

Sustaining Members of the Acoustical Society of America

The Acoustical Society is grateful for the financial assistance being given by the Sustaining Members listed below and invites applications for sustaining membership from other individuals or corporations who are interested in the welfare of the Society.

Application for membership may be made to the Executive Director of the Society and is subject to the approval of the Executive Council. Dues of $1000.00 for small businesses (annual gross below $100 million) and $2000.00 for large businesses (annual gross above $100 million or staff of commensurate size) include a subscription to the *Journal* as well as a yearly membership certificate suitable for framing. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year.

Additional information and application forms may be obtained from Elaine Moran, Office Manager, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300. Telephone: (516) 576-2360; E-mail: Elaine@acousticalsociety.org

Acentech Incorporated
www.acentech.com
Cambridge, Massachusetts
Consultants in Acoustics, Audiovisual and Vibration

ACO Pacific Inc.
www.acopacific.com
Belmont, California
Measurement Microphones, the ACOustic Interface™ System

Applied Physical Sciences Corp.
www.aphysci.com
Groton, Connecticut
Acoustics, Hydrodynamics, Electromagnetics, Manufacturing

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Cambridge, Massachusetts
R&D company providing custom advanced research based solutions

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Producer of Aircraft and Aerospace Products

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Framingham, Massachusetts
Loudspeaker Systems for Sound Reinforcement and Reproduction

D'Addario & Company, Inc.
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Farmingdale, New York
D'Addario strings for musical instruments, Evans drumheads, Rico woodwind reeds and Planet Waves accessories

G.R.A.S.
Sound & Vibration ApS
www.gras.dk
Vedbaek, Denmark
Measurement microphones, Intensity probes, Calibrators

Industrial Acoustics Company
www.industrialacoustics.com
Bronx, New York
Research, Engineering and Manufacturing–Products and Services for Noise Control and Acoustically Conditioned Environments

InfoComm International Standards
www.infocomm.org
Fairfax, Virginia
Advancing Audiovisual Communications Globally

International Business Machines Corporation
www.ibm.com/us/
Yorktown Heights, New York
Manufacturer of Business Machines

JBL Professional
www.jblpro.com
Northridge, California
Loudspeakers and Transducers of All Types

Knowles Electronics, Inc.
www.knowlesinc.com
Itasca, Illinois
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Berkeley, California
Manufacture Loudspeakers and Acoustical Test Equipment

National Council of Acoustical Consultants
www.ncac.com
Indianapolis, Indiana
An Association of Independent Firms Consulting in Acoustics

Raytheon Company
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Portsmouth, Rhode Island
Sonar Systems and Oceanographic Instrumentation: R&D in Underwater Sound Propagation and Signal Processing

Science Applications International Corporation
Acoustic and Marine Systems Operation
Arlington, Virginia
Underwater Acoustics; Signal Processing; Physical Oceanography; Hydrographic Surveys; Seismology; Undersea and Seismic Systems

Shure Incorporated
www.shure.com
Niles, Illinois
Design, development, and manufacture of cabled and wireless microphones for broadcasting, professional recording, sound reinforcement, mobile communications, and voice input–output applications; audio circuitry equipment; high fidelity phonograph cartridges and styli: automatic mixing systems; and related audio components and accessories. The firm was founded in 1925.

Sperian Hearing Protection, LLC
www.howardleight.com
San Diego, California
Howard Leight hearing protection, intelligent protection for military environments, in-ear dosimetry, real-world verification of attenuation, and education supported by the NVLAP-accredited Howard Leight Acoustical Testing Laboratory

Thales Underwater Systems
www.tms-sonar.com
Somerset, United Kingdom
Prime contract management, customer support services, sonar design and production, masts and communications systems design and production

3M Occupational Health & Environmental Safety Division
www.3m.com/occsafety
Minneapolis, Minnesota
Products for personal and environmental safety, featuring E·A·R and Peltor brand hearing protection and fit testing, Quest measurement instrumentation, audiological devices, materials for control of noise, vibration, and mechanical energy, and the E·A·RCALSM laboratory for research, development, and education, NVLAP-accredited since 1992.
Hearing conservation resource center
www.e-a-r.com/hearingconservation

Wenger Corporation
www.wengercorp.com
Owatonna, Minnesota
Design and Manufacturing of Architectural
Acoustical Products including Absorbers, Diffusers, Modular Sound Isolating Practice Rooms, Acoustical Shells and Clouds for Music Rehearsal and Performance Spaces

Wyle Laboratories
www.wyle.com/services/arc.html
Arlington, Virginia
The Wyle Acoustics Group provides a wide range of professional services focused on acoustics, vibration, and their allied technologies, including services to the aviation industry

ACOUSTICAL · SOCIETY · OF · AMERICA

APPLICATION FOR SUSTAINING MEMBERSHIP

The Bylaws provide that any person, corporation, or organization contributing annual dues as fixed by the Executive Council shall be eligible for election to Sustaining Membership in the Society.

Dues have been fixed by the Executive Council as follows: $1000 for small businesses (annual gross below $100 million); $2000 for large businesses (annual gross above $100 million or staff of commensurate size). Dues include one year subscription to *The Journal of the Acoustical Society of America* and programs of Meetings of the Society. Please do not send dues with application. Small businesses may choose not to receive a subscription to the *Journal* at reduced dues of $500/year. If elected, you will be billed.

Name of Company ______________________________

Address ______________________________

Size of Business: [ ] Small business [ ] Small business—No Journal [ ] Large business

Type of Business ______________________________

Please enclose a copy of your organization's brochure.

In listing of Sustaining Members in the Journal we should like to indicate our products or services as follows:

____
(please do not exceed fifty characters)

Name of company representative to whom journal should be sent:

____

It is understood that a Sustaining Member will not use the membership for promotional purposes.

Signature of company representatives making application:

____

Please send completed applications to: Executive Director, Acoustical Society of America, 1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, (516) 576-2360

MEMBERSHIP INFORMATION AND APPLICATION INSTRUCTIONS

Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of the application; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

BENEFITS OF MEMBERSHIP	Full Member	Associate	ce-Associate	Student
JASA Online–Vol. 1 (1929) to present	*	*	*	*
JASA tables of contents e-mail alerts	*	*	*	*
JASA, printed or CD ROM	*	*		
JASA Express Letters–online	*	*	*	*
Acoustics Today–the quarterly magazine	*	*	*	*
Proceedings of Meetings on Acoustics	*	*	*	*
Noise Control and *Sound*, *It's Uses and Control*–online archival magazines	*	*	*	*
Acoustics Research Letters Online (ARLO)–online archive	*	*	*	*
Programs for Meetings	Online	Online	Online	Online
Meeting Calls for Papers	Online	Online	Online	Online
Reduced Meeting Registration Fees	*	*		*
5 Free ASA standards per year-download only	*	*		*
Standards Discounts	*	*		*
Society Membership Directory	Online	Online	Online	Online
Electronic Announcements	*	*	*	*
Physics Today	*	*	*	*
Eligibility to vote and hold office in ASA	*			
Eligibility to be elected Fellow	*	*		
Participation in ASA Committees	*	*	*	*

QUALIFICATIONS FOR EACH GRADE OF MEMBERSHIP AND ANNUAL DUES

Student: Any student interested in acoustics who is enrolled in an accredited college or university for half time or more (at least eight semester hours). Dues: $45 per year.
Associate: Any individual interested in acoustics. Dues: $95 per year. After five years, the dues of an Associate increase to that of a full Member.
Corresponding Electronic Associate: Any individual residing in a developing country who wishes to have access to ASA's online publications only including *The Journal of the Acoustical Society of America* and Meeting Programs [see http://acousticalsociety.org/membership/membership_and_benefits]. Dues $45 per year.
Member: Any person active in acoustics, who has an academic degree in acoustics or in a closely related field or who has had the equivalent of an academic degree in scientific or professional experience in acoustics, shall be eligible for election to Membership in the Society. A nonmember applying for full Membership will automatically be made an interim Associate Member, and must submit $95 with the application for the first year's dues. Election to full Membership may require six months or more for processing; dues as a full Member will be billed for subsequent years.

JOURNAL OPTIONS AND COSTS FOR FULL MEMBERS AND ASSOCIATE MEMBERS ONLY

- ONLINE JOURNAL. All members will receive access to the *The Journal of the Acoustical Society of America (JASA)* at no charge in addition to dues.
- PRINT JOURNAL. Twelve monthly issues of *The Journal of the Acoustical Society of America*. **Cost: $35 in addition to dues.**
- CD-ROM. The CD ROM mailed bimonthly. This option includes all of the material published in the Journal on CD ROM. **Cost: $35 in addition to dues.**
- COMBINATION OF THE CD-ROM AND PRINTED JOURNAL. The CD-ROM mailed bimonthly and the printed journal mailed monthly. **Cost: $70 in addition to dues.**
- **EFFECTIVE DATE OF MEMBERSHIP**. If your application for membership and dues payment are received by 15 September, your membership and Journal subscription will begin during the current year and you will receive all back issues for the year. If you select the print journal option. If your application is received after 15 September, however, your dues payment will be applied to the following year and your Journal subscription will begin the following year.

OVERSEAS AIR DELIVERY OF JOURNALS

Members outside North, South, and Central America can choose to have print journals sent by air freight at a cost of $165 in addition to dues. JASA on CD-ROM is sent by air mail at no charge in addition to dues.

ACOUSTICAL SOCIETY OF AMERICA

1305 Walt Whitman Road, Suite 300, Melville, NY 11747-4300, asa@aip.org

For Office Use Only
Dues Rcvd ________
Aprvd by Ed ________
Aprvd by EC ________

APPLICATION FOR MEMBERSHIP

Applicants may apply for one of four grades of membership, depending on their qualifications: Student Member, Associate Member, Corresponding Electronic Associate Member or full Member. To apply for Student Membership, fill out Parts I and II of this form; to apply for Associate, Corresponding Electronic Associate, or full Membership, or to transfer to these grades, fill out Parts I and III.

PART I. TO BE COMPLETED BY ALL APPLICANTS (Please print or type all entries)

CHECK ONE BOX IN EACH COLUMN ON THE RIGHT

☐ NON-MEMBER APPLYING FOR:
☐ MEMBER REQUESTING TRANSFER TO:

☐ STUDENT MEMBERSHIP
☐ ASSOCIATE MEMBERSHIP
☐ CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP
☐ FULL MEMBERSHIP

Note that your choice of journal option *may* increase or decrease the amount you must remit.

SELECT JOURNAL OPTION:

Student members will automatically receive access to The Journal of the Acoustical Society of America online at no charge in addition to dues. Remit $45. (Note: Student members may also receive the Journal on CD ROM at an additional charge of $35.)

Corresponding Electronic Associate Members will automatically receive access to The Journal of the Acoustical Society of America and Meeting Programs online at no charge in addition to dues. Remit $45.

Applicants for **Associate or full Membership** must select one Journal option from those listed below. Note that your selection of journal option determines the amount you must remit.

[ ] Online access only—$95
[ ] Online access plus print Journal $130
[ ] Online access plus CD ROM—$130
[ ] Online access plus print Journal and CD ROM combination—$165

Applications received after 15 September: Membership and Journal subscriptions begin the following year.

OPTIONAL AIR DELIVERY: Applicants from outside North, South, and Central America may choose air freight delivery of print journals for an additional charge of $165. If you wish to receive journals by air, remit the additional amount owed with your dues. JASA on CD-ROM is sent by air mail at no charge in addition to dues.

LAST NAME	FIRST NAME	MIDDLE INITIAL	MS/MR/MRS/DR/PROF
HOME ADDRESS (STREET & NUMBER)			
CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
NAME OF ORGANIZATION OR BUSINESS			
DEPARTMENT			
ORGANIZATION ADDRESS (STREET & NUMBER)			
CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
BUSINESS TELEPHONE: AREA CODE/NUMBER	MOBILE PHONE: AREA CODE/NUMBER	HOME TELEPHONE: AREA CODE/NUMBER	
E-MAIL ADDRESS: (PRINT CLEARLY)			
DATE AND PLACE OF BIRTH (Req'd for Awards and Emeritus Status)	SEX: ☐ Female ☐ Male		
HIGHEST ACADEMIC DEGREE	DATE OF DEGREE	FIELD	INSTITUTION GRANTING DEGREE
OTHER DEGREE	DATE OF DEGREE	FIELD	INSTITUTION GRANTING DEGREE

CHECK PERFERRED ADDRESS FOR MAIL: ☐ HOME ☐ ORGANIZATION

Part I Continued ➛

PART I CONTINUED: ACOUSTICAL AREAS OF INTEREST TO APPLICANT. Indicate your three main areas of interest below, using 1 for your main interest, 2 for your second, and 3 for your third interest. (DO NOT USE CHECK MARKS.)

- ☐ ACOUSTICAL OCEANOGRAPHY **M**
- ☐ ANIMAL BIOACOUSTICS **L**
- ☐ ARCHITECTURAL ACOUSTICS **A**
- ☐ BIOMEDICAL ACOUSTICS **K**
- ☐ ENGINEERING ACOUSTICS **B**
- ☐ MUSICAL ACOUSTICS **C**
- ☐ NOISE & NOISE CONTROL **D**
- ☐ PHYSICAL ACOUSTICS **E**
- ☐ PSYCHOLOGICAL & PHYSIOLOGICAL ACOUSTICS **F**
- ☐ SIGNAL PROCESSING IN ACOUSTICS **N**
- ☐ SPEECH COMMUNICATION **H**
- ☐ STRUCTURAL ACOUSTICS & VIBRATION **G**
- ☐ UNDERWATER ACOUSTICS **J**

PART II: APPLICATION FOR STUDENT MEMBERSHIP

NAME AND ADDRESS OF COLLEGE OR UNIVERSITY WHERE PRESENTLY ENROLLED		
DEGREE EXPECTED	MONTH & YEAR DEGREE EXPECTED	NUMBER OF SEMESTER HOURS ATTENDED THIS SEMESTER
PRINT NAMES & E-MAIL ADDRESSES OF TWO FACULTY MEMBERS CERTIFYING THAT YOU ARE REGISTERED FOR AT LEAST ONE-HALF OF FULL TIME		
SIGNATURES OF THE TWO FACULTY MEMBERS LISTED ABOVE CERTIFYING THAT YOU ARE REGISTERED AT LEAST HALF TIME		
SIGNATURE OF APPLICANT		DATE

PART III: APPLICATION FOR ASSOCIATE MEMBERSHIP, CORRESPONDING ELECTRONIC ASSOCIATE MEMBERSHIP OR FULL MEMBERSHIP (and interim Associate Membership)

SUMMARIZE YOUR MAJOR PROFESSIONAL EXPERIENCE on the lines below: list employers, duties and position titles, and dates, beginning with your present position. Attach additional sheets if more space is required.

CONTRIBUTIONS TO ACOUSTICS: LIST MAIN PUBLICATIONS, PATENTS, ETC. Attach separate sheets if required.

SPONSORS AND REFERENCES: An application for full Membership requires the names, addresses, and signatures of two references who must be **full Members or Fellows** of the Acoustical Society. Names and signatures are NOT required for Associate Membership, Corresponding Electronic Associate Membership or Student Membership applications.

PRINT NAME OF REFERENCE (required for Full Member applications only)	PRINT NAME OF SECOND REFERENCE (required for Full Member applications only)
ADDRESS OF REFERENCE	ADDRESS OF SECOND REFERENCE
SIGNATURE OF REFERENCE (required for Full Member applications only)	SIGNATURE OF SECOND REFERENCE (required for Full Member applications only)
SIGNATURE OF APPLICANT	DATE

MAIL THIS COMPLETED APPLICATION, WITH APPROPRIATE PAYMENT TO: ACOUSTICAL SOCIETY OF AMERICA, 1305 WALT WHITMAN ROAD, SUITE 300, MELVILLE, NY 11747-4300.

METHOD OF PAYMENT

☐ Check or money order enclosed for $ __________ (U.S. funds/drawn on U.S. bank)

☐ American Express ☐ VISA ☐ MasterCard Signature ______________________________

(Credit card orders must be signed)

Account Number ☐☐☐☐☐☐☐☐☐☐☐☐☐☐☐☐

Expiration Date Mo. ☐☐ Yr. ☐☐

Security Code ☐☐☐☐

Due to security risks and Payment Card Industry (PCI) data security standards e-mail is NOT an acceptable way to transmit credit card information. Please return this form by Fax (631-923-2875) or by postal mail.

Regional Chapters and Student Chapters

Anyone interested in becoming a member of a regional chapter or in learning if a meeting of the chapter will be held while he/she is in the local area of the chapter, either permanently or on travel, is welcome to contact the appropriate chapter representative. Contact information is listed below for each chapter representative.

Anyone interested in organizing a regional chapter in an area not covered by any of the chapters below is invited to contact the Cochairs of the Committee on Regional Chapters for information and assistance: Catherine Rogers, University of South Florida, Tampa, FL, crogers@cas.usf.edu and Evelyn M. Hoglund, Ohio State University, Columbus, OH 43204, hoglund1@osu.edu

AUSTIN STUDENT CHAPTER
Benjamin C. Treweek
10000 Burnet Rd.
Austin, TX 78758
Email: btreweek@utexas.edu

BRIGHAM YOUNG UNIVERSITY STUDENT CHAPTER
Kent L. Gee
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Brigham Young Univ.
N283 ESC
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Email: kentgee@byu.edu
www.acoustics.byu.edu

CHICAGO
Scott D. Pfeiffer
Threshold Acoustics LLC
141 W. Jackson Blvd.
Chicago, IL 60604
Email: spreiffer@thresholdacoustics.com

UNIVERSITY OF CINCINNATI STUDENT CHAPTER
Kyle T. Rich
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Univ. of Cincinnati
231 Albert Sabin Way
Cincinnati, OH 45267
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COLUMBIA COLLEGE CHICAGO STUDENT CHAPTER
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Tallahassee, FL 32306-1200
Email: richard.morris@cci.fsu.edu

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Portsmouth, NH 03801
Email: ereuter@reuterassociates.com

UNIVERSITY OF HARTFORD STUDENT CHAPTER
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UNIVERSITY OF KANSAS STUDENT CHAPTER
Robert C. Coffeen
Univ. of Kansas
School of Architecture, Design, and Planning
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1465 Jayhawk Blvd.
Lawrence, KS 66045
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LOS ANGELES
Neil A. Shaw
www.asala.org

METROPOLITAN NEW YORK
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Riedel Audio Acoustics
443 Potter Blvd.
Brightwaters, NY 11718
Email: riedelaudio@optonline.net

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PO Box 12-1022
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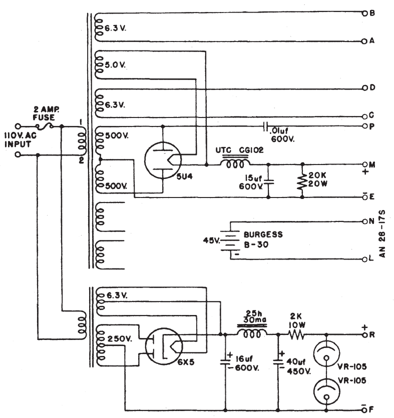

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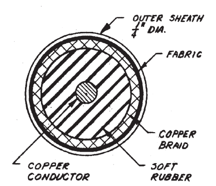

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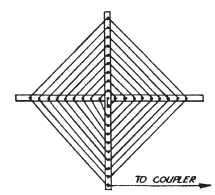

Fig.6. Configuration of the spiral mounting for the cable.

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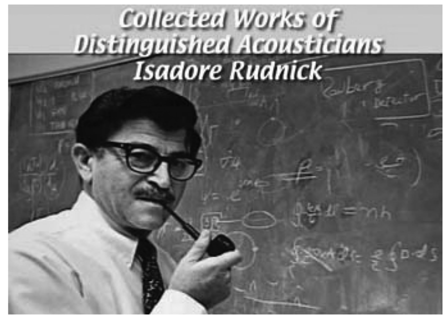
Collected Works of Distinguished Acousticians

Isadore Rudnick



The first in this series of the Collected Works of Distinguished Acousticians is that of Isadore Rudnick (May 8, 1917 - August 22, 1997). Rudnick was honored by the Acoustical Society of America (ASA) with the R. Bruce Lindsay (Biennial) Award in 1948, the Silver Medal in Physical Acoustics in 1975, and the Gold Medal in 1982. He was recognized for his acoustics research in low temperature physics with this field's most prestigious award, the Fritz London Memorial Award, in 1981 and was inducted into the National Academy of Science in 1983. Izzy's research in physical acoustics addressed boundary propagation, reciprocity calibration, high intensity sound and its biological effects, nonlinear sound propagation, and acoustics in superconductors and superfluids, including critical phenomena in bulk and thin films. The first disc in this three disc set contains reprints of Rudnick's papers from scientific journals, including 26 from the Journal of the Acoustical Society of America, and 87 from other prestigious journals, as well as some consulting reports and invited papers presented at international meetings which would otherwise be difficult to obtain. The second disc includes a montage of photographs of Rudnick with colleagues and family, Rudnick's prize winning film "The Unusual Properties of Liquid Helium", and a video of the Plenary session at the ASA's 100th meeting where Rudnick presented 90 minutes of unique and stage-sized acoustics demonstrations. While videotaped under poor conditions and of lamentable quality, the reprocessed video of acoustics demonstrations is one of the most valuable parts of this collection. The third disc is a video recording of the Memorial Session held at the 135th meeting of the ASA, which provides a comprehensive summary of Rudnick's contributions as described by former students and collaborators.

The CD was compiled by Julian D. Maynard and Steven L. Garrett of the Pennsylvania State University, State College, Pennsylvania.

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