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5–9 May 2014

Table of Contents on p. A5

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CONTENTS

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TECHNICAL PROGRAM SUMMARY

***Indicates Special Session**

Monday morning

*1aAA	Soundscape Methods and Case Studies in Architectural Projects Applications of Soundscape Techniques to the Realization of Perceived High Quality Sonic Environments in Architectural Projects
*1aAB	Comparative Perspectives on the Cocktail Party Problem I
*1aAO	Using Acoustics to Study Fish Distribution and Behavior I
*1aBA	Breast Ultrasound I
*1aED	Tools for Teaching Advanced Acoustics
*1aPP	Temporal Processing; Pitch and Timbre Perception; Speech Perception (Poster Session)
1aUW	Acoustic Tomography and Interferometry

Monday afternoon

1pAA	Room Acoustics Prediction and Measurement
*1pABa	Comparative Perspectives on the Cocktail Party Problem II
*1pABb	Dynamics of Biosonar: Time-Dependent Adaptations in Transmission, Reception, and Representation in Echolocating Animals I
*1pAO	Using Acoustics to Study Fish Distribution and Behavior II
*1pBA	Breast Ultrasound II
*1pID	Introduction to Technical Committee Research and Activities: Especially for Students and First-Time Meeting Attendees
1pMU	Topics in Musical Acoustics
*1pNS	Soundscapes: Decisions on Measurement Procedures
1pPP	From Protection to Perception
*1pSA	Undersea Vehicle Noise
1pSC	Methods and Models for Speech Communication (Poster Session)
1pSP	Sonar and Underwater Acoustic Communications

Monday evening

*1eID	Tutorial Lecture on Sound Reproduction: Science in the Service of Art

Tuesday morning

*2aAA	Uncertainty in Describing Room Acoustics Properties I
*2aAB	Dynamics of Biosonar: Time-Dependent Adaptations in Transmission, Reception, and Representation in Echolocating Animals II
*2aBA	Brain Therapy and Imaging
*2aEA	Session in Honor of Kim Benjamin
*2aMU	Where are They Now? Past Student Paper Award Winners Report
*2aNSa	Session in Honor of Kenneth Eldred
*2aNSb	Session in Honor of Harvey Hubbard
*2aPA	Beyond Basic Crystals: Viscoelastic and Piezoelectric Materials
*2aPP	Temporal Processing, Compression, and Cochlear Implants: Session in Honor of Sid P. Bacon
*2aSA	Acoustic Metamaterials I
2aSC	Speech Perception I (Poster Session)
*2aSP	Session in Honor of William M. Carey I

Tuesday afternoon

*2pAAa	Uncertainty in Describing Room Acoustics Properties II
*2pAAb	Student Design Competition
*2pAB	Acoustics as a Tool for Population Structure I
2pEA	Transduction
*2pMU	Acoustics of the Organ
*2pNS	Acoustics During Construction
*2pPA	Demonstrations in Acoustics
*2pPP	Scientific Catalyst, Collaborator, and Gadfly: Honoring the Contributions of Tino (Constantine) Trahiotis to the Understanding of Binaural Auditory Processing
*2pSA	Acoustic Metamaterials II
*2pSC	Determinants of Speech Perception: A Session in Honor of Joanne L. Miller
*2pSP	Session in Honor of William M. Carey II

Wednesday morning

*3aAAa	J. Christopher Jaffe–His Life in Acoustics
*3aAAb	Listening to the "Virtual Paul's Cross" – Auralizing 17th Century London I
3aAB	Sound Production and Reception by Animals
3aBAa	Measurement and Imaging
3BAb	Best Paper Award Poster Session
*3aED	Hands-On Acoustics Demonstrations for Middle- and High-School Students
*3aID	Future of Acoustics
*3aNS	Wind Turbine Noise
*3aPA	Acoustical Methods and Sensors for Challenging Environments
*3aPPa	Diagnostics of the Pathological Middle Ear by Wideband Acoustic Impedance/Reflectance Measures
3aPPb	Binaural Processing and Spatial Perception (Poster Session)
*3aPPc	Auditory Neuroscience Prize Lecture
*3aSAa	Session in Honor of Murray Strasberg
*3aSAb	Environmental Vibration
3aSC	Topics in Speech Production (Poster Session)
*3aSP	Intelligent Feature Selection Methods for Machine Learning Problems in Acoustics
3aUWa	Acoustic Signal and Noise Propagation and Scattering
3aUWb	Underwater Acoustics and Acoustical Oceanography Poster Session

Wednesday afternoon

*3pAAa	The Technical Committee on Architectural Acoustics Vern O. Knudsen Distinguished Lecture
*3pAAb	Listening to the "Virtual Paul's Cross" – Auralizing 17th Century London II
*3pAO	Acoustical Oceanography Prize Lecture
3pBA	Nonlinear Response of Encapsulated Microbubbles
*3pID	Hot Topics in Acoustics
3pPA	Topics in Nonlinear Acoustics
*3pSA	Acoustics of Sports
3pSC	Developmental Topics in Speech Communication

Wednesday evening

*3eED	Listen Up and Get Involved

Thursday morning

*4aAA	Green Building Acoustics Design and Challenges
*4aAB	Acoustics as a Tool for Population Structure II
*4aBA	Biomedical Applications of Low Intensity Ultrasound I
*4aEA	Session in Honor of Stanley Ehrlich
*4aID	Effective Communication Between Acoustics Professionals and the Media
*4aNS	Community Noise
4aPA	Acoustic Radiation Forces, Streaming and Applications
*4aPP	Cambridge Contributions to Auditory Science: The Moore-Patterson Legacy
*4aSA	Acoustics of Cylindrical Shells I
4aSC	Cross-Language Topics in Speech Communication (Poster Session)
*4aSP	Sensor Array Signal Processing I
*4aUW	Acoustic Vector Sensor Measurements: Basic Properties of the Intensity Vector Field and Applications I

Thursday afternoon

*4pAA	Psychoacoustics in Rooms I
*4pAB	Acoustics as a Tool for Population Structure III
*4pBAa	Biomedical Applications of Low Intensity Ultrasound II
4pBAb	Modeling and Characterization of Biomedical Systems
4pEA	Devices and Flow Noise
*4pMUa	Automatic Musical Accompaniment Systems
*4pMUb	Automatic Accompaniment Demonstration Concert
*4pNS	Out on a Limb and Other Topics in Noise
4pPA	Topics in Wave Propagation and Noise
*4pPP	Role of Medial Olivicochlear Efferents in Auditory Function
*4pSA	Acoustics of Cylindrical Shells II
4pSC	Special Populations and Clinical Considerations
*4pSP	Sensor Array Signal Processing II
*4pUW	Acoustic Vector Sensor Measurements: Basic Properties of the Intensity Vector Field and Applications II

Friday morning

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

	MONDAY, 5 MAY			TUESDAY, 6 MAY			WEDNESDAY, 7 MAY			THURSDAY, 8 MAY			FRIDAY, 9 MAY	
ROOM	MORN	AFT	EVE	MORN	AFT	EVE	MORN	AFT	EVE	MORN	AFT	EVE	MORN	AFT
Ballroom A	1aPP 8:00	1pSC 1:00		2aSC 8:00			3aPPb 8:00 3aSC 8:00							
Ballroom B	1aAB 8:00	1pABa 1:00		2aPP 8:30	2pPP 1:25		3aPPa 8:00 3aPPc 11:00			4aPP 8:00	4pAA 1:00		5aAAa 8:00	
Ballroom C	1aED 8:30	1pMU 1:30		2aMU 9:40	2pMU 1:00		3aBAb 10:30			4aID 10:00	4pMUa 2:00 4pMUb 4:45	TCMU 7:30		
Ballroom D		1pID 2:00	1eID 7:00				3aUWa 8:00	3pSC 1:00			4pSC 1:30	TCSC 7:30		
Ballroom E	1aBA 7:55	1pBA 1:00		2aBA 8:00			3aBAa 8:00	3pBA 1:00		4aBA 9:00	4pBAa 1:00 4pBAb 3:00	TCBA 7:30		
East Prefunction										4aSC 8:00			5aPP 8:00 5aSC 8:00	
550A/B				2aEA 9:00	2pEA 1:30	TCEA 4:30	3aAAb 9:00	3pAAb 1:30		4aEA 8:15	4pEA 1:30			
551A/B				2aPA 7:55	2pPA 12:55	TCPA 7:30	3aPA 7:55	3pPA 12:55		4aPA 8:30	4pPA 1:00		5aPA 9:00	
552A	1aAO 7:55	1pAO 1:00				TCAO 7:30	3aSAb 8:25							
552B		1pSP 1:30					3aSP 9:00						5aSP 8:30	
552AB				2aSP 8:25	2pSP 1:30					4aSP 8:00	4pSP 1:30			
553A/B		1pSA 1:25		2aSA 8:20	2pSA 1:00	TCSA 8:00	3aSAa 8:25	3pSA 1:25		4aSA 8:55	4pSA 1:30		5aSA 9:00	
554A/B		1pABb 3:15		2aAB 8:00	2pAB 1:00	TCPP 7:30	3aAB 9:00	3pAO 1:00		4aAB 8:00	4pAB 1:15	TCAB 7:30	5aAB 8:55	
555A/B	1aAA 8:05	1pAA 1:00		2aAA 8:15	2pAAa 1:05	TCAA 7:30	3aAAa 8:25	3pAAa 1:30	TCSP 7:30	4aAA 8:10	4pPP 1:30		5aAAb 8:00	
556A/B	1aUW 10:00	1pPP 1:00			2pAAb 1:00		3aUWb 9:00			4aUW 8:20	4pUW 1:30	TCUW 7:30	5aUW 8:00	
557		1pNS 1:00		2aNSa 8:55 2aNSb 10:35	2pNS 1:00		3aNS 8:15 3aID 11:00	3pID 1:30		4aNS 8:30	4pNS 1;30	TCNS 7:30	5aNS 8:45	
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Omni Waterplace					2pSC 1:00									

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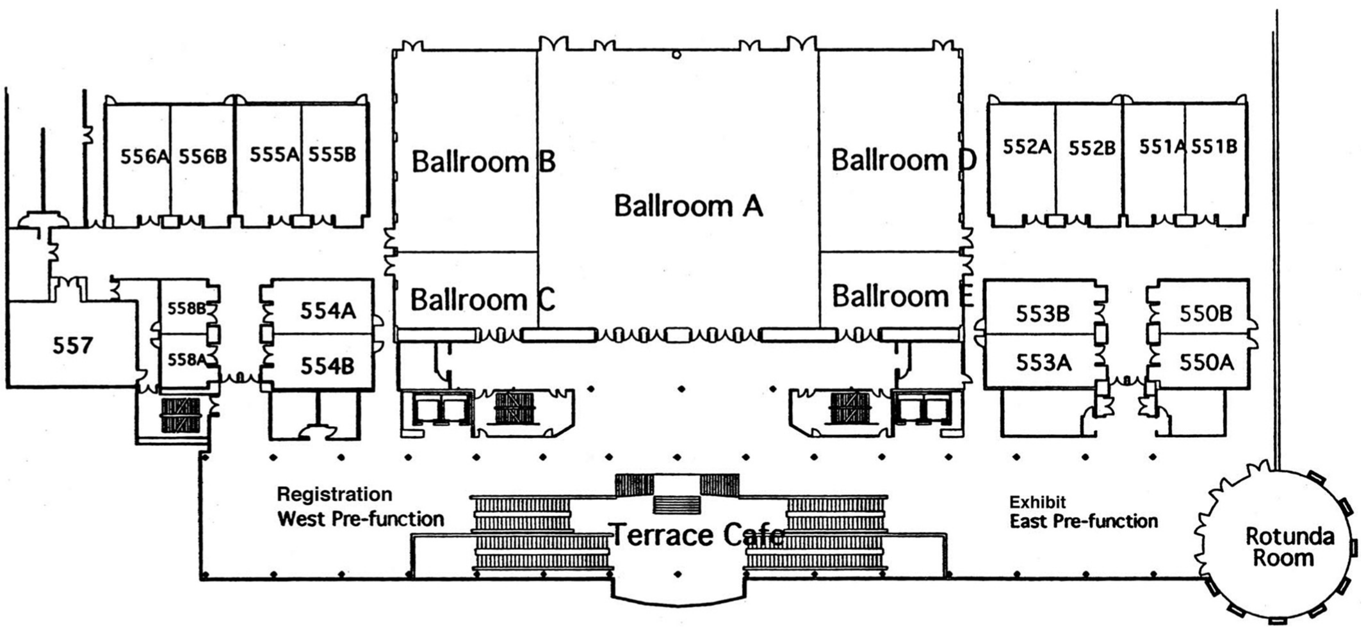

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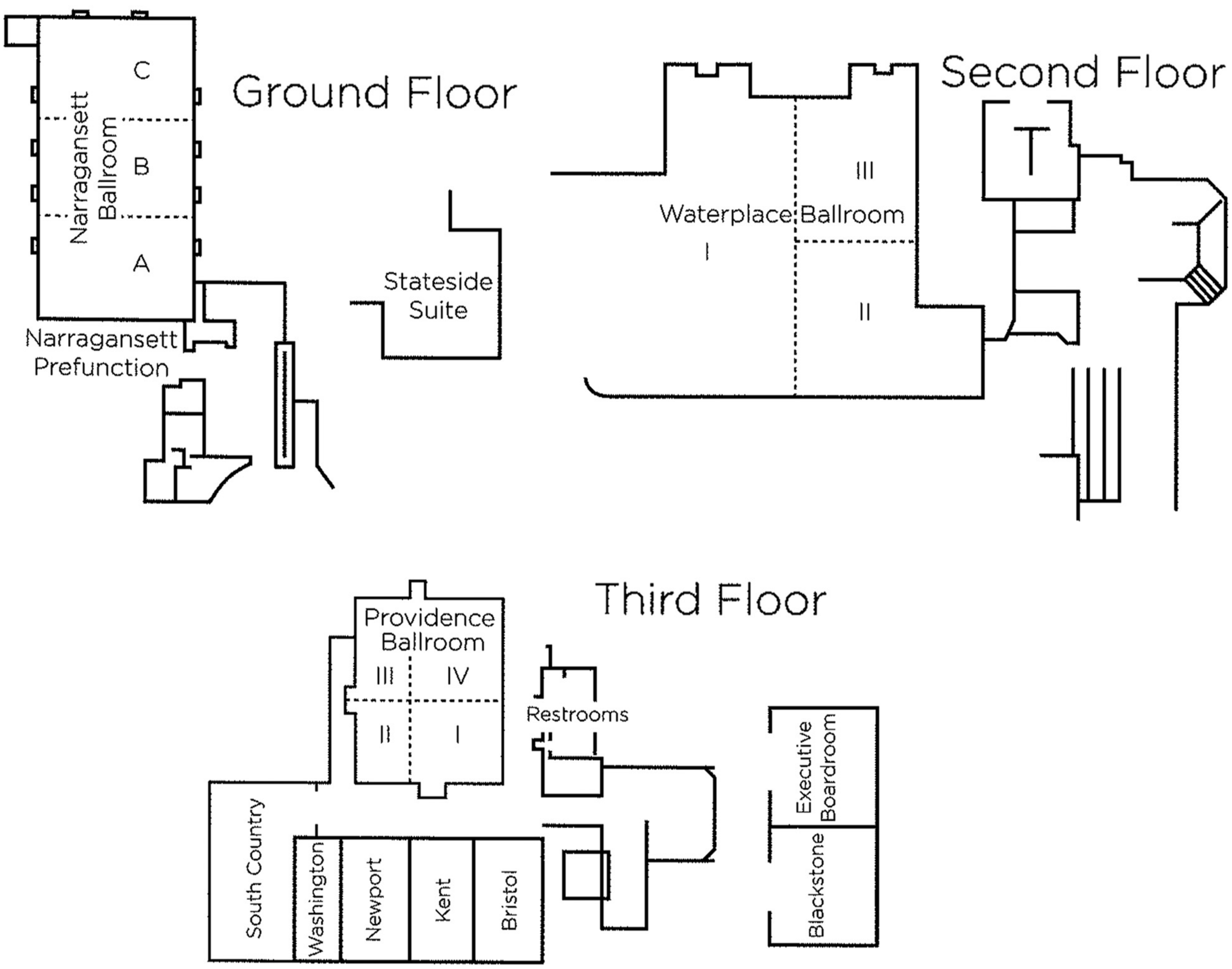

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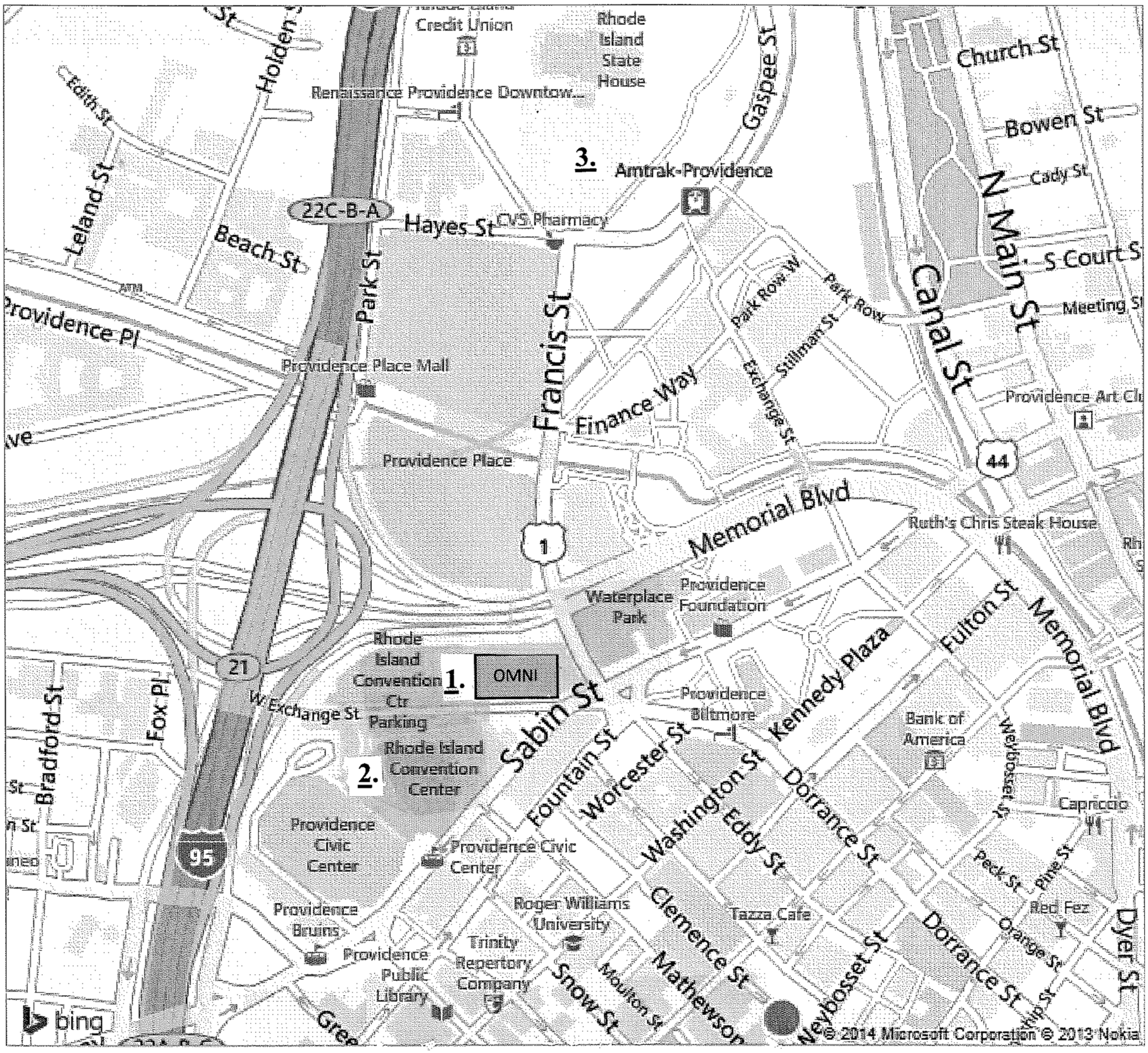

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TECHNICAL PROGRAM CALENDAR
167th Meeting
Providence, Rhode Island
5–9 May 2014

MONDAY MORNING

8:05 1aAA **Architectural Acoustics and Noise**: Soundscape Methods and Case Studies in Architectural Projects Applications of Soundscape Techniques to the Realization of Perceived High Quality Sonic Environments in Architectural Projects. 555 A/B

8:00 1aAB **Animal Bioacoustics and Psychological and Physiological Acoustics**: Comparative Perspectives on the Cocktail Party Problem I. Ballroom B

7:55 1aAO **Acoustical Oceanography and Signal Processing in Acoustics**: Using Acoustics to Study Fish Distribution and Behavior I. 552 A

7:55 1aBA **Biomedical Acoustics**: Breast Ultrasound I. Ballroom E

8:30 1aED **Education in Acoustics and Physical Acoustics**: Tools for Teaching Advanced Acoustics. Ballroom C

8:00 1aPP **Psychological and Physiological Acoustics:** Temporal Processing; Pitch and Timbre Perception; Speech Perception (Poster Session). Ballroom A

10:00 1aUW **Underwater Acoustics**: Acoustic Tomography and Interferometry. 556 A/B

MONDAY AFTERNOON

1:00 1pAA **Architectural Acoustics**: Room Acoustics Prediction and Measurement. 555 A/B

1:00 1pABa **Animal Bioacoustics and Psychological and Physiological Acoustics**: Comparative Perspectives on the Cocktail Party Problem II. Ballroom B

3:15 1pABb **Animal Bioacoustics and Signal Processing in Acoustics**: Dynamics of Biosonar: Time-Dependent Adaptations in Transmission, Reception, and Representation in Echolocating Animals I. 554 A/B

1:00 1pAO **Acoustical Oceanography and Signal Processing in Acoustics**: Using Acoustics to Study Fish Distribution and Behavior II. 552 A

1:00 1pBA **Biomedical Acoustics**: Breast Ultrasound II. Ballroom E

2:00 1pID **Interdisciplinary Student Council**: Introduction to Technical Committee Research and Activities: Especially for Students and First-Time Meeting Attendees. Ballroom D

1:30 1pMU **Musical Acoustics**: Topics in Musical Acoustics. Ballroom C

1:00 1pNS **Noise, Architectural Acoustics, and ASA Committee on Standards**: Soundscapes: Decisions on Measurement Procedures. 557

1:00 1pPP **Psychological and Physiological Acoustics**: From Protection to Perception. 556 A/B

1:25 1pSA **Structural Acoustics and Vibration and Underwater Acoustics**: Undersea Vehicle Noise. 553 A/B

1:00 1pSC **Speech Communication**: Methods and Models for Speech Communication (Poster Session). Ballroom A

1:30 1pSP **Signal Processing in Acoustics**: Sonar and Underwater Acoustic Communications. 552 B

MONDAY EVENING

7:00 1eID **Interdisciplinary**: Tutorial Lecture on Sound Reproduction: Science in the Service of Art. Ballroom D

TUESDAY MORNING

8:15 2aAA **Architectural Acoustics**: Uncertainty in Describing Room Acoustics Properties I. 555 A/B

8:00 2aAB **Animal Bioacoustics and Signal Processing in Acoustics**: Dynamics of Biosonar: Time-Dependent Adaptations in Transmission, Reception, and Representation in Echolocating Animals II. 554 A/B

8:00 2aBA **Biomedical Acoustics**: Brain Therapy and Imaging. Ballroom E

9:00 2aEA **Engineering Acoustics**: Session in Honor of Kim Benjamin. 550 A/B

9:40 2aMU **Musical Acoustics**: Where are They Now? Past Student Paper Award Winners Report. Ballroom C

8:55 2aNSa **Noise**: Session in Honor of Kenneth Eldred. 557

10:35 2aNSb **Noise**: Session in Honor of Harvey Hubbard. 557

7:55 2aPA **Physical Acoustics**: Beyond Basic Crystals: Viscoelastic and Piezoelectric Materials. 551 A/B

8:30 2aPP **Psychological and Physiological Acoustics**: Temporal Processing, Compression, and Cochlear Implants: Session in Honor of Sid P. Bacon. Ballroom B

8:20 2aSA **Structural Acoustics and Vibration, Physical Acoustics, Engineering Acoustics, and Noise**: Acoustic Metamaterials I. 553 A/B

8:00 2aSC **Speech Communication**: Speech Perception I (Poster Session). Ballroom A

8:25 2aSP **Signal Processing in Acoustics**: Session in Honor of William M. Carey I. 552 A/B

TUESDAY AFTERNOON

1:05 2pAAa **Architectural Acoustics**: Uncertainty in Describing Room Acoustics Properties II. 555 A/B

1:00 2pAAb **Architectural Acoustics, Robert Bradford Newman Student Award Fund and National Council of Acoustical Consultants**: Student Design Competition. 556 A/B

1:00 2pAB **Animal Bioacoustics**: Acoustics as a Tool for Population Structure I. 554 A/B

1:30 2pEA **Engineering Acoustics**: Transduction. 550 A/B

1:05 2pMU **Musical Acoustics**: Acoustics of the Organ. Ballroom C

1:00 2pNS **Noise and Architectural Acoustics**: Acoustics During Construction. 557

12:55 2pPA **Physical Acoustics and Education in Acoustics**: Demonstrations in Acoustics. 551 A/B

1:25 2pPP **Psychological and Physiological Acoustics**: Scientific Catalyst, Collaborator, and Gadfly: Honoring the Contributions of Tino (Constantine) Trahiotis to the Understanding of Binaural Auditory Processing. Ballroom B

1:00 2pSA **Structural Acoustics and Vibration, Physical Acoustics, Engineering Acoustics, and Noise**: Acoustic Metamaterials II. 553 A/B

1:00 2pSC **Speech Communication**: Determinants of Speech Perception: Session in Honor of Joanne L. Miller. Omni Waterplace

1:30 2pSP **Signal Processing in Acoustics**: Session in Honor of William M. Carey II. 552 A/B

WEDNESDAY MORNING

8:25 3aAAa **Architectural Acoustics**: J. Christopher Jaffe–His Life in Acoustics. 555 A/B

9:00 3aAAb **Architectural Acoustics**: Listening to the “Virtual Paul’s Cross” – Auralizing 17th Century London I. 550 A/B

9:00 3aAB **Animal Bioacoustics**: Sound Production and Reception by Animals. 554 A/B

8:00 3aBAa **Biomedical Acoustics**: Measurement and Imaging. Ballroom E

10:30 3aBAb **Biomedical Acoustics**: Best Paper Award Poster Session. Ballroom C

10:00 3aED **Education in Acoustics**: Hands-On Acoustics Demonstrations for Middle- and High-School Students. Omni Narragansett A/B

11:00 3aID **Interdisciplinary**: Future of Acoustics. 557

8:15 3aNS **Noise, Structural Acoustics and Vibration, and ASA Committee on Standards**: Wind Turbine Noise. 557

7:55 3aPA **Physical Acoustics**: Acoustical Methods and Sensors for Challenging Environments. 551 A/B

8:00 3aPPa **Psychological and Physiological Acoustics**: Diagnostics of the Pathological Middle Ear by Wideband Acoustic Impedance/Reflectance Measures. Ballroom B

8:00 3aPPb **Psychological and Physiological Acoustics**: Binaural Processing and Spatial Perception (Poster Session). Ballroom A

11:00 3aPPc **Psychological and Physiological Acoustics**: Auditory Neuroscience Prize Lecture. Ballroom B

8:25 3aSAa **Structural Acoustics and Vibration, Underwater Acoustics, and Physical Acoustics**: Session in Honor of Murray Strasberg. 553 A/B

8:25 3aSAb **Structural Acoustics and Vibration and Noise**: Environmental Vibration. 552 A

8:00 3aSC **Speech Communication**: Topics in Speech Production (Poster Session). Ballroom A

9:00 3aSP **Signal Processing in Acoustics**: Intelligent Feature Selection Methods for Machine Learning Problems in Acoustics. 552 B

8:00 3aUWa **Underwater Acoustics**: Acoustic Signal and Noise Propagation and Scattering. Ballroom D

9:00 3aUWb **Underwater Acoustics and Acoustical Oceanography**: Underwater Acoustics and Acoustical Oceanography Poster Session. 556 A/B

WEDNESDAY AFTERNOON

1:30 3pAAa **Architectural Acoustics**: The Technical Committee on Architectural Acoustics Vern O. Knudsen Distinguished Lecture. 555 A/B

1:30 3pAAb **Architectural Acoustics**: Listening to the “Virtual Paul’s Cross” – Auralizing 17th Century London II. 550 A/B

1:00 3pAO **Acoustical Oceanography**: Acoustical Oceanography Prize Lecture. 554 A/B

1:00 3pBA **Biomedical Acoustics**: Nonlinear Response of Encapsulated Microbubbles. Ballroom E

1:30 3pID **Interdisciplinary**: Hot Topics in Acoustics. 557

12:55 3pPA **Physical Acoustics**: Topics in Nonlinear Acoustics. 551 A/B

1:25 3pSA **Structural Acoustics and Vibration, Noise, and Architectural Acoustics**: Acoustics of Sports. 553 A/B

1:00 3pSC **Speech Communication**: Developmental Topics in Speech Communication. Ballroom D

WEDNESDAY EVENING

6:00 3eED **Education in Acoustics and Women in Acoustics**: Listen Up and Get Involved. Omni Narragansett A/B

THURSDAY MORNING

8:10 4aAA **Architectural Acoustics**: Green Building Acoustics Design and Challenges. 555 A/B

8:00 4aAB **Animal Bioacoustics**: Acoustics as a Tool for Population Structure II. 554 A/B

9:00 4aBA **Biomedical Acoustics**: Biomedical Applications of Low Intensity Ultrasound I. Ballroom E

8:15 4aEA **Engineering Acoustics**: Session in Honor of Stanley Ehrlich. 550 A/B

10:00 4aID **Interdisciplinary Public Relations Committee and Education in Acoustics**: Effective Communication Between Acoustics Professionals and the Media. Ballroom C

8:30 4aNS **Noise and ASA Committee on Standards**: Community Noise. 557

8:30 4aPA **Physical Acoustics**: Acoustic Radiation Forces, Streaming and Applications. 551 A/B

8:00 4aPP **Psychological and Physiological Acoustics and Speech Communication**: Cambridge Contributions to Auditory Science: The Moore-Patterson Legacy. Ballroom B

8:55 4aSA **Structural Acoustics and Vibration and Physical Acoustics**: Acoustics of Cylindrical Shells I. 553 A/B

8:00 4aSC **Speech Communication**: Cross-Language Topics in Speech Communication (Poster Session). East Prefunction

8:00 4aSP **Signal Processing in Acoustics and Underwater Acoustics**: Sensor Array Signal Processing I. 552 A/B

8:20 4aUW **Underwater Acoustics**: Acoustic Vector Sensor Measurements: Basic Properties of the Intensity Vector Field and Applications I. 556 A/B

THURSDAY AFTERNOON

1:00 4pAA **Architectural Acoustics and Psychological and Physiological Acoustics**: Psychoacoustics in Rooms I. Ballroom B

1:15 4pAB **Animal Bioacoustics**: Acoustics as a Tool for Population Structure III. 554 A/B

1:00 4pBAa **Biomedical Acoustics**: Biomedical Applications of Low Intensity Ultrasound II. Ballroom E

3:00 4pBAb **Biomedical Acoustics**: Modeling and Characterization of Biomedical Systems. Ballroom E

1:30 4pEA **Engineering Acoustics**: Devices and Flow Noise. 550 A/B

2:00 4pMUa **Musical Acoustics**: Automatic Musical Accompaniment Systems. Ballroom C

4:45 4pMUb **Musical Acoustics**: Automatic Accompaniment Demonstration Concert. Ballroom C

1:30 4pNS **Noise**: Out on a Limb and Other Topics in Noise. 557

1:00 4pPA **Physical Acoustics**: Topics in Wave Propagation and Noise. 551 A/B

1:30 4pPP **Psychological and Physiological Acoustics**: Role of Medial Olivicochlear Efferents in Auditory Function. 555 A/B

1:30 4pSA **Structural Acoustics and Vibration and Physical Acoustics**: Acoustics of Cylindrical Shells II. 553 A/B

1:30 4pSC **Speech Communication**: Special Populations and Clinical Considerations. Ballroom D

1:30 4pSP **Signal Processing in Acoustics and Underwater Acoustics**: Sensor Array Signal Processing II. 552 A/B

1:30 4pUW **Underwater Acoustics**: Acoustic Vector Sensor Measurements: Basic Properties of the Intensity Vector Field and Applications II. 556 A/B

FRIDAY MORNING

8:00 5aAAa **Architectural Acoustics and Psychological and Physiological Acoustics**: Psychoacoustics in Rooms II. Ballroom B

8:00 5aAAb **Architectural Acoustics**: Exploring the 2014 Sound and Vibration Guidelines and Case Studies for Healthcare Facilities. 555 A/B

8:55 5aAB **Animal Bioacoustics and Education in Acoustics**: Communicating the Science of Underwater Sound. 554 A/B

8:45 5aNS **Noise**: Aircraft and Fan Noise and Analysis. 557

9:00 5aPA **Physical Acoustics**: General Topics in Physical Acoustics. 551 A/B

8:00 5aPP **Psychological and Physiological Acoustics**: Potpourri (Poster Session). East Prefunction

9:00 5aSA **Structural Acoustics and Vibration**: Recent Advances in Structural Acoustics and Vibrations. 553 A/B

8:00 5aSC **Speech Communication**: Speech Perception II (Poster Session). East Prefunction

8:30 5aSP **Signal Processing in Acoustics**: Signal Processing Models for Sound Production and Perception. 552B

8:00 5aUW **Underwater Acoustics**: Underwater Acoustic Propagation. 556 A/B

SCHEDULE OF COMMITTEE MEETINGS AND OTHER EVENTS

Events in the Rhode Island Convention Center are noted with (RICC). All others will be held in the Omni.

ASA COUNCIL AND ADMINISTRATIVE COMMITTEES

Mon, 5 May, 7:30 a.m.	Executive Council	South Country
Mon, 5 May, 3:30 p.m.	Technical Council	South Country
Tue, 6 May, 7:00 a.m.	ASA Press Editorial Board	Bristol
Tue, 6 May, 7:30 a.m.	Panel on Public Policy	Newport
Tue, 6 May, 11:45 a.m.	Editorial Board	Rotunda (RICC)
Tue, 6 May, 12:00 noon	Activity Kit	Executive Boardroom
Tue, 6 May, 12:00 noon	Audit	Washington
Tue, 6 May, 12:00 noon	Prizes & Special Fellowships	Kent
Tue, 6 May, 12:00 noon	Student Council	Newport
Tue, 6 May, 1:30 p.m.	Meetings	Bristol
Tue, 6 May, 4:00 p.m.	Books+	Executive Boardroom
Tue, 6 May, 4:00 p.m.	Education in Acoustics	553 A/B (RICC)
Tue, 6 May, 4:30 p.m.	Newman Student Award	Kent
Tue, 6 May, 5:00 p.m.	Women in Acoustics	Bristol
Wed, 7 May, 6:45 a.m.	International Research & Education	Washington
Wed, 7 May, 7:00 a.m.	College of Fellows	Newport
Wed, 7 May, 7:00 a.m.	Publication Policy	Kent
Wed, 7 May, 7:00 a.m.	Regional Chapters	Bristol
Wed, 7 May, 11:00 a.m.	Medals and Awards	Washington
Wed, 7 May, 11:15 a.m.	Public Relations	Newport
Wed, 7 May, 12:00 noon	Membership	Bristol
Wed, 7 May, 1:30 p.m.	AS Foundation Board	Executive Boardroom
Wed, 7 May, 1:30 p.m.	Wind Turbine Subcommittee	Kent
Thu, 8 May, 7:00 a.m.	Archives & History	Bristol
Thu, 8 May, 7:00 a.m.	POMA	Washington
Thu, 8 May, 7:30 a.m.	Investments	Newport
Thu, 8 May, 7:30 a.m.	Tutorials	Kent
Thu, 8 May, 11:30 a.m.	Acoustics Today Advisory	Bristol
Thu, 8 May, 2:00 p.m.	Editor Search	Bristol
Thu, 8 May, 4:30 p.m.	External Affairs	Bristol
Thu, 8 May, 4:30 p.m.	Internal Affairs	Kent
Fri, 9 May, 7:00 a.m.	Technical Council	South Country
Fri, 9 May, 11:00 a.m.	Executive Council	South Country

TECHNICAL COMMITTEEE OPEN MEETINGS

Tue, 6 May, 4:30 p.m.	Engineering Acoustics	550AB (RICC)
Tue, 6 May, 7:30 p.m.	Acoustical Oceanography	552AB (RICC)
Tue, 6 May, 7:30 p.m.	Architectural Acoustics	555AB (RICC)
Tue, 6 May, 7:30 p.m.	Physical Acoustics	551AB (RICC)
Tue, 6 May, 7:30 p.m.	Psychological and Physiological Acoustics	554AB (RICC)
Tue, 6 May, 8:00 p.m.	Structural Acoustics and Vibration	553AB (RICC)
Wed, 7 May, 7:00 p.m.	Signal Processing in Acoustics	555AB (RICC)
Thu, 8 May, 7:30 p.m.	Animal Bioacoustics	554AB (RICC)
Thu, 8 May, 7:30 p.m.	Biomedical Acoustics	Ballroom E (RICC)
Thu, 8 May, 7:30 p.m.	Musical Acoustics	Ballroom C (RICC)
Thu, 7 May, 7:30 p.m.	Noise	557 (RICC)
Thu, 7 May, 7:30 p.m.	Speech Communication	Ballroom D (RICC)
Thu, 7 May, 7:30 p.m.	Underwater Acoustics	556AB (RICC)

STANDARDS COMMITTEES AND WORKING GROUPS

Mon, 5 May, 5:15 p.m.	S1, Acoustics	Kent
Mon, 5 May, 7:00 p.m.	ASACOS Steering	Bristol
Tue, 6 May, 7:00 a.m.	ASACOS	Providence 1
Tue, 6 May, 9:15 a.m.	Standards Plenary	Providence 1
Tue, 6 May, 11:00 a.m.	S12, Noise	Providence 1
Tue, 6 May, 2:00 p.m.	S3, Bioacoustics	Providence 1
Tue, 6 May 3:45 p.m.	S3/SC1, Animal Bioacoustics	Providence 1
Thu, 7 May, 1:00 p.m.	S12/WG15- Community Noise	Newport
Thu, 7 May, 2:00 p.m.	S12/WG56 - Sound-scapes in Parks	Washington

MEEETING SERVICES, SPECIAL EVENTS, SOCIAL EVENTS

Mon-Thu, 5-8 May 7:30 a.m. - 5:00 p.m.	Registration	West Prefunction, RICC
Fri, 9 May, 7:30 a.m. - 12:00 noon		
Mon-Thu, 5-8 May, 7:00 a.m. - 5:00 p.m.	E-mail Room	558B (RICC)
Fri, 9 May, 7:00 a.m. - 12:00 noon		
Mon-Thu, 5-8 May 7:00 a.m. - 5:00 p.m.	Internet Café	West Prefunction (RICC)
Fri, 9 May 7:00 a.m. - 12:00 noon		
Mon-Thu, 5-8 May 7:00 a.m.- 5:00 p.m.	A/V Preview	558B (RICC)
Fri, 9 May 7:00 a.m. - 12:00 noon		
Mon-Thu, 5-8 May 8:00 a.m. to 10:00 a.m.	Accompanying Persons	Blackstone
Sun, 4 May 1:00 p.m. - 5:00 p.m.	Short Course	Providence 1 (Omni)
Mon, 5 May 8:30 a.m. - 12:30 p.m.		
Mon, 5 May 5:30 p.m. - 6:45 p.m.	Exhibit Reception	East Prefunction (RICC)
Tue, 6 May, 9:00 a.m. - 5:00 p.m.	Exhibit	East Prefunction (RICC)
Wed, 7 May, 9:00 a.m. - 12:00 p.m.		East Prefunction (RICC)
Mon-Thu, 5-8 May 9:45 a.m. - 10:30 a.m.	Coffee Break	East Prefunction (RICC)
Fri, 9 May, 9:45 a.m. - 10:30 a.m.	Coffee Break	Rotunda (RICC)
Tue, 6 May, 3:00 p.m. - 3:30 p.m.	P.M. Exhibit Coffee Break	East Prefunction (RICC)
Mon, 5 May 7:45 a.m.	NUWC Tour	Omni Lobby
Mon, 5 May 5:00 p.m. - 5:30 p.m.	New Student Orientation	550 A/B (RICC)
Mon, 5 May 5:30 p.m. - 6:45 p.m.	Student Meet and Greet	Routunda (RICC)
Tue, 6 May, 6:00 p.m. - 7:30 p.m.	Social Hour	Ballroom A/D/E (RICC)
Wed, 7 May, 11:30 a.m. - 1:30 p.m.	Womens Luncheon	Rotunda (RICC)
Wed, 7 May, 3:30 p.m. - 5:00 p.m.	Plenary Session/Awards Ceremony	Ballroom B/C (RICC)
Wed, 7 May, 6:30 p.m. - 8:00 p.m.	Student Reception	Narragansett C (Omni)
Wed, 7 May, 7:00 p.m.	Organ Tour and Concert	Cathedral of Saints Peter and Paul
Wed, 7 May, 8:00 p.m. - 12:00 midnight	ASA Jam	Rotunda (RICC)
Thu, 7 May, 12:00 noon - 2:00 p.m.	Society Luncheon and Lecture	Narragansett A/B
Thu, 7 May, 6:00 p.m. - 7:30 p.m.	Social Hour	Ballroom A (RICC)

167th Meeting of the Acoustical Society of America

The 167th meeting of the Acoustical Society of America will be held Monday through Friday, 5–9 May 2014 at the Rhode Island Convention Center and the OMNI Providence Hotel, Providence, Rhode Island, USA.

SECTION HEADINGS

1. HOTEL INFORMATION

The OMNI Providence Hotel is the headquarters hotel where most meeting events will be held.

The cut-off date for reserving rooms at special rates has passed. Please contact the OMNI Providence Hotel for reservation information: One West Exchange Street, Providence, RI 02903; T: 401-598-8000; Fax: 401-598-8200.

2. TRANSPORTATION AND TRAVEL DIRECTIONS

Air Transportation

Located just 15 minutes from downtown Providence, Warwick's T.F. Green Airport (PVD) was recently named one of the top five airports in the US by Travel + Leisure magazine. Just off Exit 13 on Interstate Route 95, Green Airport is accessible to Boston, Cape Cod and Southeastern New England, and is fast-becoming a popular alternative to Boston's Logan Airport. For flight information visit, http://www.pvdairport.com/.

Ground Transportation

Transportation from the T. F. Green Airport to The Omni Providence Hotel and downtown Providence area hotels:

INFORMATION. The Information Booth, located at the sailboat in the airport main common area, serves many needs of the traveler to T. F. Green Airport, including maps, brochures, courtesy paging, parking and directions to locations in the local area. The staff can also provide general information on what is at T. F. Green. The information booth is staffed during normal business hours by Johnson and Wales hospitality students.

RAIL SERVICE. Providence is located on Amtrak's Northeast Corridor between Washington DC/New York City and Boston. High speed Acela Express train service transports passengers from New York City to Providence in about two and a half hours. The Massachusetts Bay Transit Authority (MBTA) runs low cost commuter trains to Providence from Boston and other points in Massachusetts. Amtrak's Providence railway station is within walking distance of The Omni Providence Hotel and The Rhode Island Convention Center. Amtrak's contact information; 1-800-USA-RAIL or visit, www.amtrak.com.

MAJOR CAR RENTAL COMPANIES. Nearly every major car rental company is represented at T.F. Green. Rental car counters are located in the Interlink building directly connected to the airport via indoor skybridge.

AIRPORT SHUTTLE SHARED-RIDE SERVICE. The shuttle departs T. F. Green Airport every hour on the hour from 5:00 a.m. to 7:00 p.m., seven days a week. It arrives and departs the Omni Providence Hotel at 17 minutes past every hour. The fee for this service is USD$11.77 per person one way or USD$23.54 round trip. Please Note: Roundtrip tickets are available for purchase inside the airport only. These tickets will guarantee a return seat. Drivers can only accept cash. At the counter, inside the airport, near baggage claims, all major credit cards are accepted. Phone 401-737-2868 or visit, www.airporttaxiri.com for more information.

TAXICABS AND LIMOUSINES. Taxis are available outside the terminal at T. F. Green Airport. Providence is approximately 10 minutes from the airport, with fares averaging USD$35.00 one way. All fares are metered. Please phone 401-737-2868 for more information.

Driving/Parking Information

Rhode Island Convention Center, One Sabin Street, Providence, RI 02903.

FROM THE NORTH. Take 95S to exit 22ABC. Merge onto 22A (Memorial Blvd.) toward Downtown. At the second light turn right onto Cookson Place. Take your next right onto West Exchange Street. Stay straight and the RI Convention Center will be directly in front of you. From the South. Take 95N to exit 22ABC. Merge onto 22A (Memorial Blvd.) toward Downtown. Turn right onto Francis Street. At the light take a right onto West Exchange Street. The RI Convention Center will be directly in front of you.

PARKING AT THE RHODE ISLAND CONVENTION CENTER. Self-Parking is available at the Rhode Island Convention Center Garage. The full day and overnight rate is USD$18.00. The RI Convention Center garage and the Omni Providence Hotel are connected via indoor walkway. http://www.riconvention.com/

Omni Providence Hotel, One West Exchange Street, Providence, Rhode Island 02903.

FROM 95 NORTH. Take Exit 22A. At the first set of lights take a right onto Francis Street, at the next set of lights make a right turn onto West Exchange Street. The hotel will be located on your right.

FROM 95 SOUTH. Take Exit 22A, at the top of the exit ramp stay in your left lane towards Downtown Providence. At the first set of lights go straight onto Memorial Boulevard. At the next set of lights make a right turn onto Exchange Street. At the stop sign make a right turn onto Exchange Terrace. At the next set of lights go straight onto West Exchange Street. The hotel will be located on your right.

FROM 295 NORTH & SOUTH. Take Exit 6 (Route 6), Follow Route 6 East to Route 10 North. Follow Route 10 to the end and bear right into Downtown Providence. At the first set of lights go straight onto Memorial Boulevard. At the next set of lights make a right turn onto Exchange Street. At the stop sign make a right turn onto Exchange Terrace. At the next set of lights go straight onto West Exchange Street. The hotel will be located on your right.

PARKING AT THE OMNI PROVIDENCE HOTEL. The hotel valet overnight and full day parking rate is USD$28.00. Self parking is available at the Rhode Island Convention Center Garage next to hotel. The self-parking rate is USD$18.00 for overnight or full day. The Rhode Island Convention Center garage and the Omni Providence Hotel are connected via indoor walkway.

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 is mailed to ASA members for each meeting and 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. REGISTRATION

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

Registration will open on Monday, 5 May, at 7:30 a.m. in the West Prefunction area, 5th floor at the Rhode Island Convention Center (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 $545 for members of the Acoustical Society of America; $645 for non-members, $150 for Emeritus members (Emeritus status pre-approved by ASA), $350 for ASA Early Career Associates (for ASA members who transferred from Student member status in 2014, 2013, or 2012), $90 for ASA Student members, $130 for students who are not members of ASA, $115 for nonmember undergraduate students, and $150 for accompanying persons.

One-day registration is available at $310 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 2014 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 $110 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 to provide verification of your status as a student and/or as an undergraduate Student.

5. ASSISTIVE LISTENING DEVICES

The ASA has purchased assistive listening devices (ALDs) for the benefit of meeting attendees who need them at technical sessions. Any attendee who will require an assistive listening device should advise the Society in advance of the meeting by writing to: Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Email: asa@aip.org

6. TECHNICAL SESSIONS

The technical program includes 105 sessions with 1145 papers scheduled for presentation during the meeting. All technical sessions will be held at the Rhode Island Convention Center.

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

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

7. TECHNICAL SESSION DESIGNATIONS

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

1-Monday, 5 May
2-Tuesday, 6 May
3-Wednesday, 7 May
4-Thursday, 8 May
5-Friday, 9 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," "b", or "c" is used to distinguish the sessions. Each paper within a session is identified by a paper number following the session-designating characters, in conventional manner. As hypothetical examples: paper 2pEA3 would be the third paper in a session on Tuesday afternoon organized by the Engineering Acoustics Technical Committee; 3pSAb5 would be the fifth paper in the second of two sessions on Wednesday afternoon sponsored by the Structural Acoustics and Vibration Technical Committee.

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

8. HOT TOPICS SESSION

The Hot Topics session 3pID will be held on Wednesday, 7 May, at 1:30 p.m. in room 557 at the Rhode Island Convention Center. Papers will be presented on current topics in the fields of Engineering Acoustics, Psychological and Physiological Acoustics, and Underwater Acoustics.

9. AUDITORY NEUROSCIENCE PRIZE LECTURE AND THE WILLIAM AND CHRISTINE HARTMANN PRIZE IN AUDITORY NEUROSCIENCE

The 2014 William and Christine Hartmann Prize in Auditory Neuroscience will be awarded to Egbert de Boer, Academic Medical Centre, Amsterdam, the Netherlands, at the Plenary Session on Wednesday, 7 May. Egbert de Boer will present the Auditory Neuroscience Prize Lecture titled "The role of physics in inner-ear physiology and auditory perception" on Wednesday, 7 May at 11:00 a.m. in Session 3aPPc in Ballroom B at the Rhode Island Convention Center.

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

The 2014 Medwin Prize in Acoustical Oceanography will be awarded to Andone Lavery, Woods Hole Oceanographic Institution, at the Plenary Session on Wednesday, 7 May. Andone C. Lavery will present the Acoustical Oceanography Prize Lecture titled "Advances in remote inference of physical and biological parameters using acoustic scattering techniques: Mapping the ocean in broadband 'color.'" on Wednesday, 7 May, at 1:00 p.m. in Session 3pAO in 554A/B at the Rhode Island Convention Center.

11. TUTORIAL LECTURE

A tutorial lecture on "Sound Reproduction: Science in the Service of Art" will be given by Floyd Toole, formerly of the National Research Council of Canada and Harman International, on Monday, 5 May, at 7:00 p.m. in Ballroom D at the Rhode Island Convention Center.

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

To defray partially the cost of the lecture a registration fee is charged. The fee is USD$25.00 and USD$7.00 for students with current ID cards.

12. SHORT COURSE

A short course on Recent Technologies for Hearing Assistance will be given on Sunday, 4 May from 1:00 p.m. to 5:00 p.m. and Monday, 5 May, from 8:30 a.m. to 12:30 p.m. in Providence I at the Omni Providence Hotel.

The objective of this short course is to provide a comprehensive overview of the current state and the perspectives of digital signal processing in hearing devices.

The short course instructors are Birger Kollmeier, PhD, MD, professor of Medical Physics at Oldenburg University in Germany and chairman of the German Cluster of Excellence "Hearing4all" and Volker Hohmann, PhD, is a professor of Applied Physics at Oldenburg University in Germany.

The registration fee is USD$300.00 (USD$125 for students) and covers attendance, instructional materials and coffee breaks. Onsite registration at the meeting will be on a space-available basis.

13. STUDENT DESIGN COMPETITION

The 2014 Student Design Competition will be displayed and judged at the Providence meeting. This competition is intended to encourage students in the disciplines of architecture, engineering, physics, and other curriculums 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 competition will be a poster session.

The Student Design Competition is sponsored by the 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.

The submitted designs will be judged by a panel of professional architects and acoustical consultants. An award of USD$1,250 will be made to the submitter(s) of the design judged "first honors." Four awards of USD$700 each will be made to the submitters of four entries judged "commendation." Entries will be on display in Session 2pAAb on Tuesday, 6 May, from 1:00 p.m. to 5:00 p.m. in 556A/B at the Rhode Island Convention Center.

14. EXHIBIT AND EXHIBIT OPENING RECEPTION

The meeting will be highlighted by an exhibit which will feature displays including computer-based instrumentation, scientific books, sound level meters, sound intensity systems, signal processing systems, devices for noise control and acoustical materials, active noise control systems and other exhibits on acoustics.

The exhibit will be held in the East Prefunction Area at the Rhode Island Convention Center adjacent to the registration area and conveniently located near the meeting rooms. It will open on Monday, 5 May, and will close on Wednesday afternoon, 7 May.

The opening reception will be held Monday, 5 May, starting at 5:30 p.m. One free drink will be provided to each registrant who attends the reception. On Tuesday exhibit hours are 9:00 a.m. to 5:00 p.m. and on Wednesday, 9:00 a.m. to 12:00 noon.

Morning and afternoon refreshments will be available in the exhibit area. Contact the Exhibit Manager for information about participating in the exhibit: Robert Finnegan, Advertising and Exhibits Division, AIP Publishing LLC, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502, Tel: 516-576-2433; Fax: 516-576-2481; E-mail: rfinnegan@aip.org.

15. TECHNICAL COMMITTEE OPEN MEETINGS

Technical Committees will hold open meetings on Tuesday, Wednesday, and Thursday evenings at the Rhode Island Convention Center. The schedule and rooms for each Committee meeting are given on page A18.

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

16. TECHNICAL TOUR TO NUWC

A tour of the acoustic test facilities at the Naval Undersea Warfare Center (NUWC) in Newport, Rhode Island will be conducted, Monday, 5 May.

A bus will leave from the main lobby near the registration desk of the Omni Providence Hotel at 1:00 p.m. Expected return time is 4:00 p.m. A fee of USD$35 will be charged to cover transportation.

A NUWC visitor request form will be forwarded to you and MUST be on file prior to the tour. Non-US citizens should register for the tour early to allow additional time for processing their visit request.

17. ORGAN TOUR AND CONCERT

A Performance and Presentation on The Organ at The Cathedral of Saints Peter and Paul, Providence, Rhode Island has been scheduled for Wednesday, 7 May 2014 at 7:00 p.m. A delightful performance and history presentation on the magnificent church organ will be provided by resident organist Phillip Faraone. The cathedral organ is a four chamber manual mechanical action Casavant, Opus 3145 designed by Larry Phelps. It is the largest mechanical action organ in North America complete with 6,616 pipes, 126 ranks and 73 stops. Many prominent organists have performed on the instrument including Dame Gillian Weir and Stephen Hamilton.

The Cathedral is a leisurely 10-minute walk from the Omni Providence Hotel and Rhode Island Convention Center http://www.cathedralprovidence.org/organ.html Cost is a USD $10 donation.

18. PLENARY SESSION AND AWARDS CEREMONY

A plenary session will be held Wednesday, 7 May, at 3:30 p.m. in Ballroom B/C.

The award ceremony will include presentations of the 2014 William and Christine Hartmann Prize in Auditory Neuroscience, the 2014 Medwin Prize in Acoustical Oceanography, the R. Bruce Lindsay Award, the Helmholtz-Rayleigh Interdisciplinary Silver Medal, and the Gold Medal. Certificates will also be presented to Fellows elected at the San Francisco meeting of the Society. See page 2317 for a list of award recipients and new fellows.

19. ANSI STANDARDS COMMITTEES

Meetings of ANSI Accredited Standards Committees and their advisory working groups will be held at the dates on times listed in the Schedule of Committee Meetings and Other Events on page A18.

Meetings of Accredited Standards Committees S1, Acoustics; S3, Bioacoustics; S3/SC1, Animal Bioacoustics, and S12, Noise, as well as the Standards Plenary meeting, are open meetings and all attendees are invited to attend and participate in the acoustical standards development process. Note that Standards Committee S2, Mechanical Vibration and Shock, will not be meeting in Providence.

Meetings of selected advisory working groups are often held in conjunction with Society meetings and are listed in the calendar 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, Suite 114E, 35 Pinelawn Road, Melville, NY 11747; T.: 631-390-0215; F: 631-390-0217; E: asastds@aip.org

20. COFFEE BREAKS

Morning coffee breaks will be held each day from 9:45 a.m. to 10:30 a.m. in the East Pre-function area from Monday through Thursday and in the Rotunda room on Friday. An afternoon break will be held on Wednesday, 3:00 p.m. to 3:30 p.m. in East Prefunction.

21. A/V PREVIEW ROOM

558B at the Rhode Island Convention Center 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 upcoming meeting of the Acoustical Society of America will have a published proceedings, and submission is optional. The proceedings will be a separate volume of the online journal, "Proceedings of Meetings on Acoustics" (POMA). This is an open access journal, so that its articles are available in pdf format 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. The format requirements for POMA are somewhat more stringent than for posting on the ASA Online Meetings Papers Site, but the two versions could be the same. The posting at the Online Meetings Papers site, however, is not archival, and posted papers will be taken down six months after the meeting. The POMA online site for submission of papers from the meeting will be opened about one-month after authors are notified that their papers have been accepted for presentation. It is not necessary to wait until after the meeting to submit one's paper to POMA. Further information regarding POMA can be found at the site http://asadl/poma/for_authors_poma. Published papers from previous meeting can be seen at the site http://asadl/poma.

23. E-MAIL ACCESS AND INTERNET CAFE

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 558B at the Rhode Island Convention Center. Wireless access will be available in the Rhode Island Convention Center and Omni Providence Hotel public areas.

24. BUFFET SOCIALS

Complimentary buffet socials with cash bar will be held on Tuesday, 6 May, and Thursday, 8 May, from 6:00 p.m. to 7:30 p.m. in Ballroom A at the Rhode Island Convention Center.

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 at 7:30 p.m. Please see page A18 for the schedule of Technical Committee meetings.

25. SOCIETY LUNCHEON AND LECTURE

The Society Luncheon and Lecture will be held on Thursday, 8 May, at 12:00 noon in Narragansett A/B at the Omni Providence Hotel. The luncheon is open to all attendees and their guests.

Tom Austin, Principal Engineer, Ocean Systems Laboratory, WHOI, will be our Society Luncheon Speaker. He will present results from field experiments in which a REMUS-100 autonomous underwater vehicle (AUV) tracked multiple tagged sharks in the open ocean over periods of several hours. The Oceanographic Systems Laboratory (OSL) developed an algorithm that allows the vehicle to use information from an active transponder to provide a three dimensional track of the animal with high spatial and high temporal resolution. Field studies were conducted in the spring and summer of 2012. Two basking sharks and four white sharks were tagged and tracked for 1-3 hours. We present the engineering developments required to create the system as well as some very exciting video footage that was featured on the Discovery Channel's "Shark Week."

Purchase your tickets at the Registration Desk before 10:00 a.m. on Wednesday, 7 May. The cost is USD$30.00 per ticket.

26. STUDENTS MEET MEMBERS FOR LUNCH

The ASA Education Committee arranges for a student to meet one-on-one with a member of the Acoustical Society over lunch. The purpose is to make it easier for students to meet and interact with members at ASA Meetings. Each lunch pairing is arranged separately. Students who are interested should contact Dr. David Blackstock, University of Texas at Austin, by email. 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 STUDENT ORIENTATION, MEET AND GREET, AND STUDENT RECEPTION

A New Students Orientation will be held from 5:00 p.m. to 5:30 p.m. on Monday, 5 May, 550A/B at the Rhode Island Convention Center. This will be followed by the Student Meet and Greet from 5:30 p.m. to 6:45 p.m. in the Rotunda at the Rhode Island Convention Center which will provide an opportunity for students to meet informally with fellow students and invited members of the Acoustical Society. Students are encouraged to attend the tutorial lecture on Sound Reproduction which begins at 7:00 p.m. in Ballroom D at the Rhode Island Convention Center.

The Students' Reception will be held on Wednesday, 7 May, from 6:30 p.m. to 8:00 p.m. in Narragansett C in the Omni Providence Hotel.

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 envelopes, for important program and meeting information pertaining only to students attending the ASA meeting. They are also encouraged to visit the official ASA Student Home Page at www.acosoc.org/student/ 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, 7 May, in the Rotunda in the Rhode Island Convention Center. Those who wish to attend must purchase their tickets in advance by 10:00 a.m. on Tuesday, 6 May. The fee is USD$30 for non-students and USD$15 for students.

29. JAM SESSION

The tradition continues! You are invited to the Rotunda in the Providence Convention Center on Wednesday night, 7 May from 8:00 p.m. to midnight for an epic 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. ACCOMPANYING PERSONS PROGRAM

Spouses and other visitors are welcome at the Providence meeting. The on-site registration fee for accompanying persons is USD$150. A hospitality room for accompanying persons will be open in the Blackstone Room at the Omni Providence Hotel from 8:00 a.m. to 10:00 a.m. Monday through Thursday where information about activities in and around Providence will be provided. A representative from the Providence and Warwick Convention and Visitors Bureau will be present on Monday and Tuesday mornings from 9:00 a.m. to 10:00 a.m. to talk about Providence and the many things to do and places to see.

31. WEATHER

April showers will have brought a bloom to May flowers and a definite Spring feel in the air. Temperatures are typically in the mid 60s during the day. Dropping to the cool low 50s at night.

32. TECHNICAL PROGRAM ORGANIZING COMMITTEE

James F. Lynch, Chair; Timothy K. Stanton, Acoustical Oceanography; James A. Simmons, Animal Bioacoustics; William Cavanaugh, Timothy J. Foulkes, Architectural Acoustics; Tyrone Porter, E. Carr Everbach, Biomedical Acoustics; David T. Bradley, Education in Acoustics; Roger Richards, Engineering Acoustics; Eric Dieckman, Musical Acoustics; Nancy Timmerman, Noise; David A. Brown, Physical Acoustics; Patrick Zurek, Richard Freyman, Jayaganesh Swaminathan, Psychological and Physiological Acoustics; John R. Buck, Signal Processing in Acoustics; Rachel M. Theodore, Mark Tiede, Robert Port, Speech Communication; Robert M. Koch, Structural Acoustics and Vibration; Cathy Ann Clark, Gopu R. Potty Underwater Acoustics.

33. MEETING ORGANIZING COMMITTEE

James H. Miller, General Chair; Gopu R. Potty, Cochair; James F. Lynch, Technical Program Chair: Andrea M. Simmons and James A. Simmons, Audio Visual; John R. Buck, Signs; Cathy Ann Clark and David A. Brown, Technical Tours; Gail Paolino, Meeting Coordinator.

34. PHOTOGRAPHING AND RECORDING

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

35. ABSTRACT ERRATA

This meeting program is Part 2 of the April 2014 issue of *The Journal of the Acoustical Society of America*. Corrections, for printer's errors only, may be submitted for publication in the Errata section of the *Journal*.

36. GUIDELINES FOR ORAL PRESENTATIONS

Preparation of Visual Aids

- See the 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.
- If you are using transparencies, all material should be within an 8x9-inch (20x23 cm) frame to ensure projection of the entire page.
- For computer presentations, use all of the available screen area using landscape orientation with very thin margins.

If your institutions logo must be included, place it at the bottom of the slide.
- Sans serif fonts (e.g., Arial, Calibri, and Helvetica) are much easier to read than serif fonts (e.g., Times New Roman) especially from afar. Avoid thin fonts (e.g., the horizontal bar of an e may be lost at low resolution thereby registering as a c.)
- Do not use underlining to emphasize text. It makes the text harder to read.
- All axes on figures should be labeled.
- No more than 3–5 major points per slide.
- Consistency across slides is desirable. Use the same background, font, font size, etc. across all slides.
- Use appropriate colors. Avoid complicated backgrounds and do not exceed four colors per slide. Backgrounds that change from dark to light and back again are difficult to read. Keep it simple.
- If using a dark background (dark blue works best), use white or yellow lettering. If you are preparing slides that may be printed to paper, a dark background is not appropriate.
- If using light backgrounds (white, off-white), use dark blue, dark brown or black lettering.
- DVDs should be in standard format.

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

37. SUGGESTIONS FOR EFFECTIVE POSTER PRESENTATIONS

Content

The poster should be centered around two or three key points supported by the title, figures, and text. The poster should be able to "stand alone." That is, it should be understandable even when you are not present to explain, discuss, and answer questions. This quality is highly desirable since you may not be present the entire time posters are on display, and when you are engaged in discussion with one person, others may want to study the poster without interrupting an ongoing dialogue.
- To meet the "stand alone" criteria, it is suggested that the poster include the following elements, as appropriate:
  - Background
  - Objective, purpose, or goal
  - Hypotheses
  - Methodology
  - Results (including data, figures, or tables)
  - Discussion
  - Implications and future research
  - References and Acknowledgment

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

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

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

- Tables should be used sparingly and presented in a simplified format.

Presentation

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

Other suggestions

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

38. GUIDELINES FOR USE OF COMPUTER PROJECTION

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

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

Introduction

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

Guidelines

Set your computer's screen resolution to 1024 × 768pixels 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, a video switcher will be available. During the question and answer period of the previous speaker, connect your laptop to the video switcher. It is good protocol to initiate your slide show (e.g., run PowerPoint) immediately once connected, so the audience doesn't have to wait. When it is your turn to present, the session chair will press the button on the switcher corresponding to the appropriate number of the input to which you connected (indicated on the cord you plugged into your computer). 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.
Macintosh

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

- Hook everything up before powering anything on. (Connect the computer to the RGB input on the projector).
- Turn the projector on and boot up the Macintosh. If this doesn't work immediately, you should make sure that your monitor resolution is set to 1024 × 768 for an XGA projector or at least 640 × 480 for an older VGA projector. (1024 × 768 will most always work.). You should also make sure that your monitor controls are set to mirroring. If it's an older powerbook, it may not have video mirroring, but something called simulscan, which is essentially the same.
- Depending upon the vintage of your Mac, you may have to reboot once it is connected to the computer projector or switcher. Hint: you can reboot while connected to the computer projector in the A/V preview room in advance of your presentation, then put your computer to sleep. Macs thus booted will retain the memory of this connection when awakened from sleep. Alternatively, you can reboot while connected to the video switcher during the previous speaker's presentation, but it is safer to queue this up in advance of the session.
- 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 fiddling with your 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 computer/video prep room. Likewise, you may have to set your laptop's resolution and color depth via the monitor's Control Panel to match that of the projector, which settings you should verify prior to your session. Linux
- Most Linux laptops have a function key marked CRT/ LCD or two symbols representing computer versus projector. Often that key toggles on and off the VGA output of the computer, but in some cases, doing so will cause the computer to crash. One fix for this is to boot up the BIOS and look for a field marked CRT/LCD (or similar). This field can be set to Both, in which case the signal to the laptop is always presented to the VGA output jack on the back of the computer. Once connected to a computer projector, the signal will appear automatically, without toggling the function key. Once you get it working, don't touch it and it should continue to work, even after reboot.

39. DATES OF FUTURE ASA MEETINGS

For further information on any ASA meeting, or to obtain instructions for the preparation and submission of meeting abstracts, contact the Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502; Telephone: 516-576-2360; Fax: 516-576-2377; E-mail: asa@aip.org

168th Meeting, Indianapolis, Indiana, 27–31 October 2014
169th Meeting, Pittsburgh, Pennsylvania, 18–22 May 2015
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

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:

Ronald L. Bannister
James F. Bartram
Raymond M. Brach
David J. Broad
Lawrence G. Copley
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Allen H. Shiner
William J. Strong
Alan D. Stuart
Alan M. Teplitzky
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Armin T. Wright

TWENTY-FIVE YEAR AWARDS

The following individuals have been members of the Society continuously for twenty-five years. They will be sent "Silver" certificates in recognition of the mutual advantages derived from their long-time association with the Society.

Katsuji Akamatsu
Victor A. Akulichev
Ronald J. Baken
Anna Barney
Ginger A. Barth
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ANNUAL GIVING TO THE ACOUSTICAL SOCIETY FOUNDATION FUND – 2013

The Acoustical Society of America Foundation Board is deeply grateful for all contributions received in 2013. To help express this gratitude, the list of donors to the Foundation Fund is published below for all donations received in 2013.

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Wright, Wayne M.

Contributors—up to $99
Abramson, Arthur S.
Akamatsu, Katsuji
Allen, John S.
Allen, Jont B.
Arai, Yasuhiko
Arau, Higini
Arveson, Paul T.
Augspurger, George L.
Avan, Paul A.
Bahtiarian, Michael
Balachandran, Balakumar
Becker, Kyle M.
Bedard, Alfred J., Jr.
Bell, Thaddeus G.
Berger, Elliott H.
Bernstein, Jared C.
Bevan, Jeffrey S.
Bishop, Dwight E.
Bissinger, George A.
Blackstock, David T.
Bond, Zinny S.
Boyce, Suzanne E.
Braslau, David
Brown, Neal A.
Brown, Steven M.
Burroughs, Courtney B.
Cable, Peter G.
Cameron, Robert H.
Campbell, Christopher S.
Campbell, Joseph P., Jr.
Campbell, Melanie M.
Campbell, Murray D.
Carome, Edward F.
Carter, G. Clifford
Chalikia, Magdalene H.
Chan, Dominic S. F.
Channel, Lawrence E.
Choi, Min Joo
Church, Charles Clair
Ciocca, Valter
Colburn, H. Steven
Cole, Bernard F.
Coleman, John S.
Colleran, C. Nicholas
Collier, Sandra Lee
Commander, Kerry W.
Cook, Rufus L.
Copley, Lawrence G.
Cottingham, James P.
Creelman, C. Douglas
Cristini, Paul
Crystal, Thomas H.
Cummings, William C.
Dalby, Jonathan M.
Das, Pankaj K.
Davies, Patricia
Davis, Donald B.
Deane, Grant B.
De Bruijn, Alex
Dembowski, James S.
Demorest, Marilyn E.
Diaz-Sanchidrian, Cesar
Di Benedetto, Maria-Gabriella
Dooley, Kevin A.
Dranetz, Abraham I.
Duifhuis, Hendrikus
Dunens, Egons K.
Edmonds, Peter D.
Ellermeier, Wolfgang
Emanuel, Floyd W.
Ensminger, Dale
Ermann, Michael
Erskine, Fred T.
Escudeiro, Roberto L.
Evans, Jack B.
Fagen, David G.
Fahnline, John B.
Fenwick, Alan J.
Ferguson, Elizabeth L.
Finch, Robert D.
Finegold, Lawrence S.
Fitzgerald, James M.
Flanagan, James L.
Florentine, Mary
Forbes, Barbara J.
Fourakis, Margaritis S.
Fullerton, Jeff
Funatsu, Seiya
Galaitsis, Anthony
Gallego, Vicente
Game, Christopher J. A.
Gammell, Paul M.
Garrelick, Joel M.
Gaumond, Charles F.
Gebbie, John T.
Geers, Thomas L.
George, Ballard W.
Germano, Carmen P.
Giacometti, Alberto
Gibian, Gary L.
Goldstein, Julius L.
Gonzalez, Roberto A.
Gorges, Heinz A.
Greenwood, Margaret S.
Guttman, Newman
Hanson, Carl
Harada, Tatsuhiko
Harris, Katherine S.
Hasse, Raymond W.
Hatlestad, Kathryn W.
Henetz, Tania K.
Herstein, Peter D.
Hillenbrand, James M.
Hirsch, Karl Wilhelm
Hobelsberger, Max
Holford, Richard L.
Horoshenkov, Kirill V.
House, Arthur S.
Huaquin, Mario A.
Hueter, Theodor F.
Hughes, Larry F.
Hughes, William
Hui, Wang
Hull, Andrew J.
Hursky, Paul
Hurst, Charles J.
Ishii, Kiyoteru
Ivey, Larry E.
Jackson, Darrell R.
Jaeger, Stephen M.
Johnson, Keith A.
Jones, Edward
Joshi, Suyash N.
Kawahara, Hideki
Keast, David N.
Kenbu, Teramoto
Kentaro, Ishizuka
Kewley-Port, Diane
Kim, Youngshin

Kimura, Tomonori
Klepper, David Lloyd
Koenig, Laura L
Korman, Murray S.
Krahe, Detlef
Krause, Nicholas D.
Krebs, Donald F.
Kreiman, Jody E.
Krokstad, Asbjorn
Kruse, Roland
Kryter, Karl D.
Kumamoto, Yoshiro
Labuda, Cecille Pemberton
Lalani, Asad
Lancey, Timothy W.
Landgren, Kenneth M.
Langley, Robin S.
Lavoie, Lisa M.
Lelong, Joel
Lerner, Armand
Letcher, Stephen V.
Levin, Franklyn K.
Levitt, Andrea G.
Levitt, Harry
Lin, Wen-Hwang
Loftis, Charles B., Jr.
Loney, Jack W.
Lutolf, John J.
Lyons, Joy E.
Lysaght, Thomas A.
MacGillivray, John R.
Mackenzie-Hoy, Terry
Maddieson, Ian
Mahoney, Robert F.
Manuel, Sharon Y.
Martin, Gordon E.
Matteson, Samuel E.
McEachern, James F.
McGregor, Stuart D.
McLaughlin, Hugh J.
Meesawat, Kittiphong
Mehl, James B.
Mellert, Volker
Memoli, Gianluca
Mershon, Donald H.
Miller, Norman D.
Milner, Paul
Minor, Michael A.
Moffett, Mark B.
Moore, James A.
Mullins, Joe H.
Murphy, David P., Jr.
Murphy, Joseph E.
Namba, Seiichiro
Neff, Donna L.
Neubert, Vernon H.
Noble, John M.
Nolle, Alfred W.
Norris, Thomas
Novarini, Jorge C.
O'Malley, Honor
Pallas, Marie-Agnes F.
Papcun, George
Patton, Richard S.
Paulauskis, John A.
Penardi, Paul A.
Pestorius, Frederick M.
Peterson, Ronald G.
Peterson, William M.
Petit, Christine
Pettersen, Michael S.
Piacsek, Andrew A.
Pierce, Allan D.
Pierson, David Michael
Port, Robert F.
Powell, Robert E.
Prout, James H.
Quartararo, Louis R.
Rayess, Nassif E.
Raymond, Jason L.
Reynolds, Duane A.
Richarz, Werner G.
Ringheim, Matias
Robert, Daniel W.
Rochat, Judith L.
Rogers, Catherine L.
Roos, Mark S.
Rosenberg, Aaron E.
Rosowski, John J.
Rubega, Robert A.
Rueckert, Daniel C.
Russo, Arlyne E.
Rutledge, Janet C.
Saathoff, Shawn D.
St. Pierre, Richard L.
Saito, Shigemi
Sakurai, Masatsugu
Sato, Takuso
Saunders, James C.
Scanlon, Michael V.
Schafer, Frederick C.
Scott, Vaughan D., Jr.
Sessler, Gerhard M.
Shattuck-Hufnagel, Stefanie R.
Shekhar, Himanshu
Shimoda, Hidemaro
Shiner, Allen H.
Short, Patrick J.
Silbiger, Herman R.
Simson, John
Smirnov, Nickolay A.
Sokolowski, Roy T.
Sone, Toshio
Spring, Neil F.
Stanton, Timothy K.
Stepinski, Tadeusz
Steppat, Michael
Stern, Richard M.
Stevens, Kenneth N.
Stewart, John L.
Stinson, Michael R.
Stumpf, Folden B.
Sutherland, Louis C.
Svet, Victor
Taylor, Joseph A., IV
Taylor, M. Martin
Temkin, Samuel
Teranishi, Arthur M.
Terts, Pearu
Testen, Joseph C.
Thunder, Thomas D.
Timmerman, Nancy S.
Towers, David A.
Tran, Thuy Ng
Tubis, Arnold
Turner, Joseph A.
Ungar, Eric E.
Unverferth, John E.
van Dommelen, Wim A.
Van Dyke, Michael B.
Veale, Edward
Vendittis, David J.
Victor, Alfred E.
Visintini, Lucio
Vorperian, Houri K.
Walker, Bruce E.
Walkling, Robert A.
Wang, Rong Ging
Webster, John C.
Weinberg, Henry
Wendelboe, Gorm
Wetherill, Ewart A.
Whalen, Doug H.
Williams, Hollis E. F.
Wilson, Douglas J.
Wold, Donald C.
Worley, Robert D.
Wright, Matthew C. M.
Wright, Richard Albert
Yahya, Iwan
Yamamoto, Teruji
Yang, Shiuh Kuang
Young, Robert F.
Zinberg, Mildred F.
Zwislocki, Jozef J.

Session 1aAA

Architectural Acoustics and Noise: Soundscape Methods and Case Studies in Architectural Projects: Applications of Soundscape Techniques to the Realization of Perceived High Quality Sonic Environments in Architectural Projects

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

Juergen Bauer, Cochair
Dept. of Architecture, Waterford Inst. of Technol., Granary, Hanover St., Waterford, Ireland

Chair's Introduction—8:05

Invited Papers

8:10

1aAA1. Soundscape and architecture—What is your vision? Bennett M. Brooks (Brooks Acoustics Corporation, 30 Lafayette Square-Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com) and Dennis Paoletti (Paoletti Consulting, San Mateo, CA)

The soundscape technique combines physical acoustical parameters with user perceptions of the sonic environment in a context of meaning. This method can be a powerful analysis and design tool for a wide range of projects, including building interior and exterior spaces, site planning, urban and transportation planning, environmental noise control, public parks, etc. The soundscape method can also be a useful marketing and project management tool, with the goal to address acoustical concerns as early as possible in the architectural design process, even in the inspiration phase. Sonic perceptions of the built environment are often a vital part of the vision for a project, and must be expressed at the outset to be fully incorporated in the design. Innovative project delivery methods and contract structures such as Integrated Project Delivery (IPD), unlike design-bid-build, assign shared risk and reward among the design, construction, and management teams. Design inputs are solicited from all stakeholders and design team members very early, before programming. This and similar delivery methods offer great opportunities for practitioners, through soundscaping, to include acoustics in the initial project discussions, and to advance the implementation of quality sonic environments. Project case study examples are discussed.

8:30

1aAA2. Architectural acoustics and sense of place. Michael A. Stocker (Ocean Conservation Res., P.O. Box 559, Lagunitas, CA 94938, mstocker@OCR.org)

A large part of the work in architectural acoustics is focused on noise control, tailoring reverberation time, improving speech intelligibility, and reduction of specular reflections. But there are other factors that play into the suitability of an acoustic environment for a specific purpose that are not easily expressed on numerical indices. In this presentation, the author reviews his own work in the US National Holocaust Museum "Daniel's House" exhibit, the Museo Papalote del Niño Rainforest Tree Exhibit in Mexico City, and various other settings in public exhibition spaces. The presentation will also review some historic successes and failures in public enclosures and soundscapes that hinge on how the user/visitor experiences their relationship with the acoustical surroundings.

8:50

1aAA3. Traffic design for soundscape improvements. Klaus Genuit and André Fiebig (HEAD Acoust. GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, klaus.genuit@head-acoustics.de)

Different kinds of traffic contribute to urban acoustic environments and therefore have a major impact on urban soundscapes. The noise of traffic depends on several aspects such as traffic management, traffic routing, traffic composition, and infrastructure. In the past, any optimization of these aspects targets only on sound pressure level reduction, neglecting perceptional relevant phenomena. It is well known that the human hearing does not work like a simple sound level meter. Besides loudness, humans perceive psychoacoustic properties of noise and notice certain sound events and sources. Thus, any improvement of traffic noise must be guided by knowledge from psychoacoustics and cognition. To sustainably improve the appraisal of a soundscape, traffic must be deliberately designed. In different research projects, the psychoacoustic potential of traffic design was systematically investigated. For example, the perceptual difference between roundabouts and intersections with and without traffic lights was investigated, psychoacoustic requirements for the layout of road markings were studied, and the required penetration level of electric cars for a substantial noise reduction beyond sound pressure level considerations was an object of investigation. Options and possibilities of traffic design from a psychoacoustic perspective and their implications for urban planning will be presented.

9:10

1aAA4. Perceived space, an essential soundscape variable. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

The perceived space within a soundscape is free from the constraints of place and time. The space defined by the architecture need not define the perceived space within a soundscape. Space is a variable, determined by the combined influences of the architecture, sounds system, source signals, and signal processing applied in the creation of the soundscape. The perceived space is also free to change over time through dynamic signal processing and any other variable acoustics.

9:30

1aAA5. Production techniques for perceptually realistic soundscape auralization. Matthew Azevedo (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, mazevedo@acentech.com)

It is now possible to build complex, parametrically accurate auralizations with many sound sources. However, parametric accuracy does not necessarily translate to perceptual accuracy. In order to create soundscapes that listeners will perceive as "real," audio production techniques are required, which generally lie outside of the expertise of an acoustician. This paper provides an overview of audio recording, mixing, and postproduction techniques that are commonly employed by music producers, sound designers, and audio engineers in creating soundtracks for television, film, and video games, which provide listeners with perceptually realistic sound experiences. These techniques are presented in the context of auralization projects in which they were employed to successfully bridge the gap between parametric and experiential accuracy, with an emphasis on methods which satisfy both. Also discussed are modeling and convolution techniques which lay the groundwork on which these recording and mixing techniques can be deployed successfully.

9:50–10:05 Break

10:05

1aAA6. Clues from the brief, from the site and from the users Architectural design and soundscape. Juergen Bauer (Dept. of Architecture, Waterford Inst. of Technol., Granary, Hanover St., Waterford, Co Waterford 00000, Ireland, jbauer@wit.ie)

This paper investigates how the Soundscape approach and the architectural design process can inform each other. Designing is an intuitive process and is therefore not bound by strict rules. However, most architects will agree that a successful design concept is guided by three factors: First, a design proposal needs to meet its purpose, i.e., to respond to the demands of a design brief. Second, a design proposal should contribute to its location and the surrounding neighborhood, whether it is blending into this context or emerging from it. Lastly, the actual idea for a design proposal is informed by the clues from the brief and the clues from the site. Supported by case studies, it is argued that the Soundscape approach can greatly contribute to the design process and its focus on the brief, the location, and the conceptual idea. It is further discussed how the public debate on architecture in the context of town planning and design competitions can be beneficial to the Soundscape approach and its interest in the contribution of the users and the local stakeholders.

10:25

1aAA7. The challenge of interdisciplinarity in soundscape. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 10587, Germany, b.schulte-fortkamp@tu-berlin.de)

Interdisciplinarity is one of the most stressed player in Soundscape. Over time when starting to collaborate in and with different disciplines, it became obvious that it is necessary to understand the needs in soundscapes as well as the use of soundscape techniques in the respective disciplines. One of the most important outcomes in the COST project on Soundscape TD 0804 with respect to... that a common language is needed to guarantee the collaboration. Especially in the field of Architecture, there is the need to understand why Soundscape and landscape have some similarities but also that landscape and soundscape are different in their focus. The paper will discuss understandings and misunderstandings in the respective fields and provide an orientation guideline.

Contributed Papers

10:45

1aAA8. Acoustic measurements based on a soundscape analysis in an open-plan classroom in a primary school. Sang Bong Shin (Architecture, Univ. of Florida, 2330 SW Williston Rd., APT534, Gainesville, FL 32608, archisangbong@gmail.com)

It is critical to fully understand the acoustic environment in open plan classrooms because there is a current increasing popularity of a new type of classrooms and because it has been reported that the open plan classroom has a serious problem of noise. Since the architectural features of the open plan classrooms are different from those of traditional classrooms, traditional measurement methods are not sufficient to investigate the new type of classrooms. In this study, a new type of open-plan classroom combined with small classrooms is examined with soundscape approaches. Acoustical events occurring in the open-plan classroom in a primary school were analyzed, and the activities that created the specific acoustical events were observed using methods of soundwalks, focus group discussions, and narrative interviews. Also acoustic measurements were conducted with measurement sets from soundscape analysis and from traditional methods in the building. The results of measurements were compared to determine the differences in the effects of measurement methods on the acoustical events in open plan classrooms. The study found that the results of the acoustical measurements based on soundscape analysis are different from those of traditional measurement methods. The differences among the measurement sets demonstrate that it is useful to use soundscape analyze to understand open plan classrooms.

11:00

1aAA9. Soundscape evaluation on Mississippi State University campus. Yalcin Yildirim (Mississippi State Univ., 103 Eudora Wwelty Dr., Avalon Apartments, D/11, Starkville, MS 39759, yy214@msstate.edu)

The term soundscape, used first time at the end of 1970s, refers to the sum of the sounds which can be heard and perceived by people in a specific environment (Schafer). The concept of soundscape has recently paid attention to planning and design disciplines where the focus point is commonly placed on the visual, rather than the acoustic aspect. The perception of an outdoor environment does not only depend on the physical features of a site, but also relies on the characteristics of the users. Thus, this research will examine how objective measurement of soundscape might be different from subjective perceptions of users in the Mississippi State University Campus as a public open space due to demographic and climatic variations. Stage one, as a pilot or a preliminary study, was a soundscape walk with a small group in four selected sites. Stage two will include more detailed interviews in these sites with a much larger sample size from the general public. At the end of the study, the research findings will help to characterize soundscapes of different types of urban open spaces and to understand how a person perceives and evaluates the sound qualities in these areas.

11:15

1aAA10. "Illuminating" reflection orders in architectural acoustics using SketchUp and light rendering. J. Parkman Carter (Architectural Acoust., Rensselaer Polytechnic Inst., 32204 Waters View Circle, Cohoes, NY 12047, cartej8@rpi.edu)

The conventional architecture workflow tends to—quite literally—"overlook" matters of sound, given that the modeling tools of architectural design are almost exclusively visual in nature. The modeling tools used by architectural acousticians, however, produce visual representations, which are, frankly, less than inspirational for the design process. This project develops a simple scheme to visualize acoustic reflection orders using light rendering in the freely available and widely used Trimble SketchUp 3D modeling software. In addition to allowing architectural designers to visualize acoustic reflections in a familiar modeling environment, this scheme also works easily with highly complex geometry. The technique and examples will be presented.

11:30–12:00 Panel Discussion

MONDAY MORNING, 5 MAY 2014 BALLROOM B, 8:00 A.M. TO 11:55 A.M.

Session 1aAB

Animal Bioacoustics and Psychological, and Physiological Acoustics: Comparative Perspectives on the Cocktail Party Problem I

Mark Bee, Cochair
Dept. of Ecology and Evolutionary Biology, Univ. of Minnesota, 100 Ecology, 1987 Upper Buford Circle, St. Paul, MN 55108

Micheal L. Dent, Cochair
Psychology, Univ. at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260

Invited Papers

8:00

1aAB1. Directional cues for sound source segregation in birds, crocodilians, and lizards. Catherine E. Carr (Biology, Univ. Maryland, Stadium Dr., 20742, College Park, MD 20742-4415, cecarr@umd.edu) and Jakob Christensen-Dalsgaard (Dept. of Biology, Univ. Southern Denmark, Odense, Denmark)

Sound source segregation depends on neural mechanisms that enhance directionality. The main directional cues are interaural time difference (ITD) and interaural level difference (ILD). Birds, crocodilians, and lizards have a brainstem circuit used for detection of ITDs. In birds and crocodilians, this circuit forms a map of ITD by delay lines and coincidence detection. The physical range of ITDs for these maps are small in animals with small heads, which should make detection of ITDs difficult. Both birds and crocodilians have coupled ears, however, which extend the range of ITDs available as well as enhancing ILD. Lizards have even more strongly coupled ears, extending the ITD range by a factor 3, but ITD and ILD covary, and a large part of the ITD is a constant delay created by filtering by interaural cavities, chiefly producing enhanced lateralization. All lizard auditory nerve fibers show strongly directional responses, and effectively every neuron in the lizard auditory pathway is directional, enhancing the already strong lateralization by simple EI-type neural processing, but with no clear maps of auditory space. Thus, the processing of sound direction in the bird, alligator, and lizard CNS is different, but all three groups have mechanisms for enhancing sound source directionality.

8:20

1aAB2. Spatial stream segregation by cats, rats, and humans. John C. Middlebrooks, Peter Bremen (Otolaryngol., Univ. of California at Irvine, Rm. 116 Medical Sci. E, Irvine, CA 92697-5310, j.midd@uci.edu), Lauren K. Javier, and Justin D. Yao (Neurobiology and Behavior, Univ. of California at Irvine, Irvine, CA)

Spatial hearing aids a listener in disentangling multiple competing sound sequences. We find that separation of around 10° between target and masker sound sources permits humans and cats to hear interleaved sound sequences as segregated streams, thus enabling a "rhythmic masking release" task requiring recognition of target rhythms. In cats and rats, neurons in primary auditory cortex (A1) exhibit spatial stream segregation in that they synchronize selectively to one of two interleaved sequences of noise burst originating from spatially separated sources. Cortical spatial selectivity is markedly sharper under competing-sound conditions compared to that observed with single sound sources. Cortical responses are predicted well by a model that incorporates moderate spatial selectivity inherited from the brainstem sharpened by forward suppression at the level of thalamocortical synapses. Consistent with that model, spatial stream segregation in rats is stronger in cortical area A1 than in the ventral division of the medial geniculate body, its principal source of thalamic input. In cats, psychophysical performance was better for high-frequency sounds, and cortical stream segregation was stronger for neurons having high characteristic frequencies. In contrast, human psychophysics was better for low-frequency sounds, suggesting that the larger heads of humans provide them with greater interaural time differences.

8:40

1aAB3. Solutions to cocktail-party-like problems in acoustic insects. Heiner Römer (Zoology, Karl-Franzens-Univ., Universitätsplatz 2, Graz 8010, Austria, heinrich.roemer@uni-graz.at)

Insects often communicate by sound in mixed species choruses; like humans and many vertebrates, in crowded social environments, they thus have to solve cocktail-party-like problems in order to ensure successful communication. This is a particular problem in species-rich environments like tropical rainforests with background noise levels of up to 60 dB SPL. I describe three “bottom-up” mechanisms in cricket receivers, which contribute to an excellent neuronal representation of conspecific signals under such conditions. First, more sharply tuned frequency selectivity of the receiver reduces the amount of masking energy around the species-specific calling song frequency, resulting in a signal-to-noise ratio (SNR) of −-8 dB, when masker and signal were broadcast from the same side. Second, spatial release from masking improved the SNR by further 6 to 9 dB. Neurophysiological experiments carried out in the nocturnal rainforest yielded a further improvement of SNRs by 8 dB compared to the laboratory. Finally, a neuronal gain control mechanism enhances the contrast between the responses to signals and the masker, by inhibition of neuronal activity in inter-stimulus intervals. The results indicate that without knowledge of receiver properties and the spatial release mechanisms the detrimental effect of noise may be strongly overestimated.

9:00

1aAB4. Cross-modal integration and non-linear relationships: What can frogs tell us about solving cocktail party problems? Ryan C. Taylor (Biology, Salisbury Univ., 1101 Camden Ave., Salisbury, MD 21801, rctaylor@salisbury.edu)

Courtship in most anuran amphibians occurs in noisy environments, analogous to human communication at cocktail parties. Female frogs express strong mating preferences for particular properties of male vocalizations, but how they identify individual callers within the noisy chorus environment remains unclear. One possible mechanism is cross-modal integration, whereby females attend to both acoustic and visual cues (male vocal sac inflation). In choice experiments, we used a robotic frog with an inflating vocal sac, combined with acoustic playbacks, to test the role of cross-modal integration in female túngara frogs. In nature, male túngara frogs produce a two-note courtship call and the vocal sac inflates synchronously during production of both notes. We tested female mating preferences when we artificially varied the temporal synchrony of the vocal sac inflation relative to the two call notes. Some combinations elicited a strong preference from females, some combinations generated a strong aversive response, and other combinations were neutral. These data show that females conduct cross-modal assessments of male callers. The temporal combinations that elicited positive, negative, or neutral responses were not predictive in a linear fashion, however, suggesting that the integration of visual cues may strongly modulate auditory perception in females.

9:20

1aAB5. Comparative perception of temporally overlapping sounds. Erikson G. Neilans and Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260, mdent@buffalo.edu)

Parsing the auditory scene is a problem faced by humans and animals alike. Characteristics such as frequency, intensity, and location all help organisms assign concurrent sounds to specific auditory objects. The timing of sounds is also important for object perception. When sounds totally overlap in time, ascribing separate sounds to individual objects is difficult. However, slight temporal separations make this task slightly easier. In humans, synchronous streams of high and low frequency tones are heard as a single auditory stream. When the tones are slightly offset in time from one another, a second stream emerges. Here, we compared the perception of simultaneous, asynchronous, and partially overlapping streams of tones, human speech sounds, and budgerigar (*Melopsittacus undulatus*) contact calls in budgerigars and humans using operant conditioning methods. Human and bird subjects identified the partially overlapping stimuli differentially. Both species required less temporal separation to identify the sounds as “asynchronous” for the complex stimuli than for the pure tones. Interestingly, the psychometric functions differed between the two species. These results suggest that both humans and nonhumans are capable of using temporal offsets for assigning auditory objects, and that the ability to do this depends on the spectrotemporal characteristics of the sounds.

9:40

1aAB6. Echolocating bats face a cocktail party nightmare when they fly together in cluttered environments. Cynthia F. Moss (Dept. of Psych. and ISR, Univ. of Maryland, Biology-Psych. Bldg. 2123M, College Park, MD 20742, cynthia.moss@gmail.com), Clement Cechetto (AGROSUP, Inst. Nationale Superieur des Sci. Agronomique, Dijon, Ce, DC France), Michaela Warnecke (Psychol. and Brain Sci., Johns Hopkins Univ., Baltimore, MD), Chen Chiu, Wei Xian, and Benjamin Falk (Dept. of Psych. and ISR, Univ. of Maryland, College Park, MD)

Echolocating bats often operate in the presence of conspecifics and in cluttered environments, which can be characterized as a “cocktail party nightmare.” Each bat’s sonar vocalization can result in an echo cascade from objects distributed in direction and range. Adding to the acoustic clutter are the signals from neighboring bats. Past studies demonstrate that bats adapt their echolocation to avoid signal jamming from conspecifics by adjusting the frequencies of their vocalizations, as well as going silent. When bats fly alone in densely cluttered environments, they adjust the frequencies of call pairs to disambiguate overlapping echo streams. How do echolocating bats adapt to both conspecific signals and clutter? We sought to answer this question by flying big brown bats in a large room equipped with high-speed video and audio recording equipment. In baseline trials, bats flew alone in an empty room and were later introduced to an artificial forest, first individually and later in pairs. The echolocation behavior and flight paths are analyzed to evaluate the spectro-temporal adjustments of bat calls and silent behavior as animals progressed from open room, to forest, to forest with conspecifics. The results shed light on how echolocating bats adapt to a "cocktail party nightmare."

10:00

1aAB7. Neural representations of the cocktail party in human auditory cortex. Jonathan Z. Simon (Biology, Univ. of Maryland, Dept. of Elec. & Comput. Eng., Univ. of Maryland, College Park, MD 20742, jzsimon@umd.edu)

An auditory scene is perceived in terms of its constituent auditory objects. Here, we investigate how auditory objects are individually represented in human auditory cortex, using magnetoencephalography (MEG) to record the neural responses of listeners. In a series of experiments, subjects selectively listen to one of two competing streams, in a variety of auditory scenes. In the acoustically richest example, subjects selectively listen to one of two competing speakers mixed into a single channel. Individual neural representations of the speech of each speaker are observed, with each being selectively phase locked to the rhythm of the corresponding speech stream, and from which can be exclusively reconstructed the temporal envelope of that speech stream. The neural representation of the attended speech, originating in posterior auditory cortex, dominates the responses. Critically, when the intensities of the attended and background speakers are separately varied over a wide intensity range, the neural representation of the attended speech adapts only to the intensity of that speaker, but not to the intensity of the background speaker. Overall, these results indicate that concurrent auditory objects, even if spectrally overlapping and not resolvable at the auditory periphery, are indeed neurally encoded individually as objects, in auditory cortex.

10:20–10:35 Break

10:35

1aAB8. Temporal and spatial coherence as cues for across-frequency grouping in treefrogs. Mark Bee (Ecology, Evolution and Behavior, Univ. of Minnesota, 100 Ecology, 1987 Upper Buford Circle, St. Paul, MN 55108, mbee@umn.edu)

Humans exploit environmental regularities in sounds to perceptually bind acoustic energy occurring simultaneously at different frequencies. Such abilities influence vowel perception in speech and timbre perception in music. Other animals solve similar binding problems in the recognition of species-specific acoustic signals. Moreover, they commonly do so using auditory systems that differ in notable ways from that of mammals. This study of two treefrog species investigated temporal and spatial coherence as cues that promote grouping of two spectral bands emphasized in their acoustic signals. In two-alternative choice tests, females preferred temporally and spatially coherent calls over alternatives in which the onsets/offsets of the two bands were time-shifted by more than 25 ms or in which the two bands were spatially separated by 7.5° or more. These results, which suggest temporal coherence and spatial coherence promote across-frequency auditory grouping, are notable given differences in how the two spectral bands are processed by the anuran auditory system. Sound energy in the high- and low-frequency bands primarily enters the auditory system via different pathways (tympanum and body wall, respectively) and is encoded primarily by different papillae in the inner ear (basilar papilla and amphibian papilla, respectively).

10:55

1aAB9. Release from auditory masking with complex signals and complex noise in the bottlenose dolphin (*Tursiops truncatus*). Brian K. Branstetter (National Marine Mammal Foundation, 2240 Shelter Island Dr. #200, San Diego, CA 92106, brian.branstetter@nmmpfoundation.org), Jennifer S. Trickey, Kimberly L. Bakhtiari, Amy Black (G2 Software Systems Inc., San Diego, CA), and James J. Finneran (US Navy Marine Mammal Program, San Diego, CA)

Dolphins are social animals that rely heavily on passive and active acoustics for communication, navigation, foraging, and detecting predators. Auditory masking, from both natural and anthropogenic noise sources, may adversely affect these fitness-related capabilities. The dolphin's ability to detect a variety of complex signals (both dolphin phonations and tonal signals) masked by Gaussian, comodulated, snapping shrimp, and ice squeaks noise was tested. Detection thresholds were measured using a go/no-go adaptive staircase procedure. Masking patterns were similar for all signals (whistles, burst-pulse, and pure tones) except for click signals. Masking from ice squeaks resulted in the largest masked thresholds, while snapping shrimp and comodulated noise resulted in a release from masking relative to thresholds from Gaussian noise. Click signals were most difficult to detect when masked by snapping shrimp. Recognition thresholds were estimated for whistle-like signals using a cross-modal, matching-to-sample procedure. Recognition thresholds were on average 4 dB greater than detection thresholds for all noise types. The auditory mechanisms governing the results are discussed. [Work supported by the ONR.]

11:15

1aAB10. Neural correlates of hearing in noise in macaque auditory cortex. Yale Cohen and Sharath Bennur (Univ. Pennsylvania, 3400 Spruce St., 5 Ravdin, Philadelphia, PA 19104, ycohen@mail.med.upenn.edu)

The perception of sound in a noisy environment is a critical function of the auditory system. Here, we describe results from our study into the link between neural activity in the auditory cortex and the hearing-in-noise tasks described above. We recorded neural activity from single neurons in the core auditory cortex (i.e., A1) while monkeys were participating in these tasks. Neural recordings were conducted with tetrodes, and the frequency of the target matched the best frequency of the recorded auditory neuron. We found that the relative intensity of the target tone in the presence of the noise masker significantly modulated the response of A1 neurons. In contrast, the presentation of the target sound alone did not elicit a significant response from A1 neurons. This suggests a task-relevant contextual modulation of A1 responses during hearing in noise. Additionally we found no correlation between the monkey's behavioral choices—as assessed by their responses on choice trials—and A1 activity. Our results suggest that the encoding of a sound of interest in the presence of a noise masker is an active process, providing new insights into the neural basis for hearing in noise in the auditory system.

11:35

1aAB11. Communicating in a cacophony: Possible solutions to the cocktail party problem in treefrog choruses. Joshua J. Schwartz (Biology and Health Sci., Pace Univ., 861 Bedford Rd., Pleasantville, NY 10570, jschwartz2@pace.edu)

Male treefrogs advertise for mates in dense assemblages characterized by high levels of noise and acoustic clutter. Non-mutually exclusive approaches to ameliorating the "cocktail party" problem in frog choruses could involve signal production or perception. Male neotropical *Dendropsophus microcephalus* employ multi-note calls and can rapidly alter inter-note timing to reduce call overlap. Adjustments are made selectively such that interference is most effectively reduced among closest neighbors. There is evidence that intensity and perhaps spatial cues contribute to this selectivity. Male gray treefrogs, *Hyla versicolor*, do not seem to exhibit selective attention in a way that reduces call interference among nearest neighbors, and changes made in call duration and rate that occur with increasing noise levels do not aid in signal detection by females. Moreover, auditory induction, by which the auditory system might perceptually restore masked or missing elements of pulsatile calls, does not seem to occur. Although, under some circumstances, differences in call frequency may help females distinguish among neighboring males, naturalistic spectral differences do not seem to help females perceptually separate the overlapping calls of such males. There is evidence, however, that spatial separation of males can contribute to signal segregation by listening females during acoustic interference.

MONDAY MORNING, 5 MAY 2014 552 A, 7:55 A.M. TO 11:50 A.M.

Session 1aAO

Acoustical Oceanography and Signal Processing in Acoustics: Using Acoustics to Study Fish Distribution and Behavior I

Kelly J. Benoit-Bird, Chair
College of Earth, Ocean & Atmos. Sci., Oregon State Univ., 104 CEOAS Admin Bldg., Corvallis, OR 97331

Timothy K. Stanton, Chair
Woods Hole Oceanogr. Inst., MS #11, Woods Hole, MA 02543-1053

Chair's Introduction—7:55

Invited Papers

8:00

1aAO1. The benefits and challenges of passive acoustic monitoring of fish. Carrie C. Wall (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 216 UCB, Boulder, CO 80309, carrie.bell@colorado.edu)

While is it widely known that numerous families of fish produce sound for communication, discerning when, where, and who is more difficult. Recent developments in passive acoustic technologies have facilitated marine bioacoustic studies to effectively monitor soniferous fishes. Because acoustic data can be collected over a wide range of habitats and depth for long periods of time, passive acoustic monitoring can map and monitor marine species to efficiently provide year-round information on distribution. This presentation reviews data recorded using moored passive acoustic arrays and hydrophone-integrated gliders. Low frequency (50–6000 Hz) sounds recorded by these methods provide a better understanding of the diurnal and spatial distribution of known fish calls (e.g., red grouper). However, this is seemingly overwhelmed by the vast number of sounds produced by unknown species. Instrument and anthropogenic noise, managing the large of amounts of data collected, and identifying the source of previously undocumented sounds, are just some of the challenges passive acoustic monitoring presents. The connection between sound and important behavior, including courtship and spawning, the application for fisheries management, and the potential impacts of aquatic noise on critical behaviors that affect populations exemplifies the need to overcome these issues.

8:20

1aAO2. Broadband classification and statistics of long-range, mid-frequency sonar measurements of aggregations of fish. Benjamin Jones (Oceanogr., Naval Postgrad. School, Monterey, CA), Timothy K. Stanton (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., MS #11, Woods Hole, MA 02543, tstanton@whoi.edu), John A. Colosi (Oceanogr., Naval Postgrad. School, Monterey, CA), Roger C. Gauss (Acoust. Div., Naval Res. Lab., Washington, DC), Joseph M. Fialkowski (Acoust. Div., Naval Res. Lab., Washington, California), and J. M. Jech (NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA)

Scattering from fish can constitute a significant portion of the high-amplitude echoes in the case of a horizontal-looking sonar system operating at mid-frequencies (1–10 kHz). In littoral environments, reverberation from fish with resonant gas-filled swimbladders can dominate bottom and surface reverberation and add spatio-temporal variability to an already complex acoustic record. Measurements of sparsely distributed, spatially compact fish aggregations have been conducted in the Gulf of Maine using a long-range, broadband sonar with continuous coverage over the frequency band of 1.5–5 kHz. Concurrent downward-looking, multi-frequency echosounder measurements (18, 38, and 120 kHz), and net samples of fish are used in conjunction with physics-based acoustic models to classify and statistically characterize the long-range fish echoes. A significant number of echoes, which are at least 15 dB above background levels, were observed in the long-range data and classified as due to mixed assemblages of swimbladder-bearing fish. These aggregations of fish produce highly non-Rayleigh distributions of echo magnitudes. The probability density functions of the echoes are accurately predicted by a computationally efficient, physics-based model that accounts for beam-pattern and waveguide effects as well as the scattering response of aggregations of fish. [Work supported by the U.S. Office of Naval Research.]

Contributed Papers

8:40

1aAO3. Getting more for less: Increasing the accessibility of water column sonar data for fisheries management. Carrie C. Wall, Charles Anderson (Cooperative Inst. for Res. in Environ. Sci., Univ. of Colorado at Boulder, 216 UCB, Boulder, CO 80309, carrie.bell@colorado.edu), and Susan J. McLean (National Geophysical Data Ctr., NOAA, Boulder, CO)

Active acoustic technology is of increasing importance for studies examining fish populations and biological abundance in the water column. Multibeam echosounders are employed routinely on NOAA fishery vessels to estimate biomass, conduct trophic- and species-level identification, measure school morphology and behavior, and characterize habitat for commercially important species. These surveys deliver valuable information for ecosystem-based fisheries management but they also produce massive amounts of data that are costly and difficult to maintain. With its ability to store and preserve large datasets, NOAA's National Geophysical Data Center is acquiring and archiving acoustic data collected from NOAA and academic fleets. Through these efforts, an accessible archive of acoustic water column data will be made available to researchers and the public around the world. A web-based search engine will allow anyone to identify where data were collected, what instrument was used, and access the raw data and associated products. Years of decreasing funding for the sciences have necessitated our ability to get more information and more users out of data currently collected. This globally accessible archive is a large step in that direction. Of most importance is identifying how best to tap the archive to benefit current and future fisheries research and management.

8:55

1aAO4. Detection of fish near the bottom with a hull-mounted multibeam echosounder. Christian de Moustier (2535 Midway Dr. #81777, San Diego, CA 92138, cpm@ieee.org)

In single-beam or split-beam fisheries echosounding, bottom echoes often mask echoes from fish hovering near the bottom. Likewise, in down-looking multibeam echosounding bottom echoes received in the main lobe of near normal incidence beams appear in the sidelobes of the other beams and obscure weaker echoes received at the same range in the mainlobe of these other beams. Some multibeam sonars use frequency division multiplexing to avoid crosstalk between beams. However, it was shown recently [de Moustier, *Proc. MTS/IEEE OCEANS'13*, San Diego, September 23–27, 2013] that detection of fish near the seafloor is possible with a multibeam echo-sounder operating at a single acoustic frequency. This is achieved with a signal processing technique based on the ordered statistic constant false alarm rate (OS-CFAR) detection method used in radar. Here, the OS-CFAR operator is applied in the angle domain, rather than the customary range domain, and it estimates the signal-to-clutter ratio across all angles at a given time slice or range increment at the output of the beamformer. In this context, clutter encompasses both clutter and noise against which the desired signal is to be detected. Data collected with a 160 kHz multibeam echosounder are used to demonstrate the technique.

9:10

1aAO5. Observations of fission/fusion processes in fish aggregations using a multibeam echosounder. Thomas C. Weber (Ctr. for Coastal and Ocean Mapping, Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, weber@ccom.unh.edu), Daniel Grunbaum (School of Oceanogr., Univ. of Washington, Seattle, WA), and Timothy K. Stanton (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA)

Models of fish behavior within aggregations typically incorporate a fish's ability to sense near neighbors. Some newer models also include a cognitive functionality that allows the fish to understand not only the action of the near neighbors but also their intent. This cognitive function might be manifested, for example, in some stochastic estimate of the movement of a local population rather than at the individual level. Inherent in such a function are the temporal and spatial scales at which a fish's cognitive function operates. To help constrain these scales we have analyzed acoustic backscatter from walleye pollock collected with a multibeam echosounder as part of the NOAA Alaska Fisheries Science Center survey. During this survey, repeat passes that were approximately 1 nmi long were collected at approximately 15 mi intervals over an aggregation of fish. In at least one case, we have been able to acoustically observe an initial group of fish that undergoes both fission (splitting) and fusion (recombining) behaviors. In doing so, we are able to track the net movement, speed, and size of various parts of the group, thereby providing some ground truth for cognitive functionality models. [Research supported by the U.S. Office of Naval Research.]

9:25

1aAO6. Phased-array Doppler sonar measurements at the equator: Currents or swimming? Jerry A. Smith (SIO, UCSD, 9500 Gilman Dr., M.S. 0213, La Jolla, CA 92093-0213, jasmith@ucsd.edu)

A 64-channel 200 kHz phased array Doppler sonar (PADS) was deployed on the Equator at 140 W, sampling a vertical slice of the ocean, from 9 m to 200 m depth by 100 degrees, twice per second. The instrument was operated for two nearly continuous time-series, Oct. 10–20 and Oct.24 to Nov. 3, 2012. While the PADS was operated off the Starboard side of the R/V Revelle, a "Fast-CTD" (FCTD) was simultaneously operated off the port stern, sampling to 250 m every 2.5 min. Headway was maintained at less than 1 m/s relative to the surface flow (which was small, in contrast to the ~1.5 m/s undercurrent at 100 to 130 m depth). Because of this slow headway, motile scatterers were able to develop ship-centric swimming patterns, particularly at night. This, in turn, introduced some subtly non-physical characteristics in the estimated vertical velocities in particular. Animations of the backscatter intensity reveal a variety of scales and speeds of the scatterers. Visual nighttime inspection revealed a preponderance of small squid (< 30 cm) with the occasional large predator passing rapidly through (not identified). In spite of this biological interference, reasonable profiles of horizontal velocity were produced via a simple de-spiking selector.

9:40

1aAO7. Impacts of the warming of the continental shelf in spring 2012 on acoustic propagation conditions and fish distributions north of Cape Hatteras. Glen Gawarkiewicz, Ke Chen, James F. Lynch (Woods Hole Oceanogr. Inst., M.S. #21, Woods Hole, MA 02543, ggawarkiewicz@whoi.edu), Thomas M. Grothues (Rutgers Univ., NB, NJ), Arthur Newhall, and Ying-Tsong Lin (Woods Hole Oceanogr. Inst., Woods Hole, MA)

During May 2012, we conducted hydrographic surveys in conjunction with studies of acoustic scattering from fish schools north of Cape Hatteras. The waters of the continental shelf were greater than 4° Degrees C. warmer than prior observations during typical spring-time conditions in May 1996. In addition, the temperature gradients which normally exist across the shelfbreak front were absent, leading to intensification of the shelfbreak frontal jet. We relate the warming to large-scale atmospheric shifts and also report on the absence of cold water fish species, which were expected to be abundant in the study area.

9:55–10:10 Break

10:10

1aAO8. Deep-diving autonomous underwater vehicle provides insights into scattering layer dynamics. Kelly J. Benoit-Bird (College of Earth, Ocean & Atmos. Sci., Oregon State Univ., 104 CEOAS Admin Bldg., Corvallis, OR 97331-0000, kbenoit@coas.oregonstate.edu)

Organisms within deep scattering layers are often too densely packed to be ensonified as individuals using surface or seafloor based sensors, are too fast to be easily captured by research nets yet too small for most fishing gear, and are mixed with other individuals, making it difficult to interpret acoustic data from these ecologically important animals. To address this, we integrated a two-frequency, split-beam echosounder into an autonomous underwater vehicle (AUV) capable of flight at 600 m. As part of a study on whale foraging ecology off the Channel Islands, California, we flew the echosounders through scattering layers found at three depth ranges. Echoes were obtained from individual scatterers within layers being foraged upon by Risso's dolphins. Examining the echo statistics throughout layers revealed remarkable heterogeneity of echo strength and frequency response within layers that generally appeared homogeneous with respect to these same characteristics from ship-based echosounders. Some layers were internally layered but most features showed distinct, small patches of similar scatterers adjacent to those with different characteristics. The extensive horizontal coverage and near target sampling permitted by the AUV-based echosounders are providing a new understanding of the scales of biological organization within horizontally extensive scattering features.

10:25

1aAO9. Mapping the scatterscape of pelagic side scan sonar targets relative to oceanographic features. Thomas M. Grothues (Inst. of Marine and Coastal Sci., Rutgers Univ., Marine Field Station, 800 c/o 132 Great Bay Blvd., Tuckerton, NJ 08087, grothues@marine.rutgers.edu), Arthur E. Newhall, James F. Lynch, Glen G. Gawarkiewicz (Woods Hole Oceanogr. Inst., Woods Hole, MA), and Kaela S. Vogel (Dept. of Marine Biology, Univ. of North Carolina, Wilmington, Wilmington, NC)

Sonar reconnaissance of fishes for stock assessment and research has been an effective and minimally invasive method of gathering abundance and distribution data on scales of 10s to 100s of km since the 1950s. Yet, classification of fishes remains one of the greatest challenges of active sonar surveys. Many variables affect sonar reflection, including size, shape, orientation to the sonar source, the spatial relationship of individuals in a school to each other, and the number and distribution of individuals within a school. The long wavelengths of low frequency (typically <60 kHz) that allow depth penetration provide poor small scale resolution for identifying objects. High frequency side scan sonar (600—900 kHz), while imaging only over short ranges, can resolve individual fish and thus orientation and behavior relevant to understanding low frequency sonar returns and ecology. We demonstrate here that autonomous underwater vehicles (AUVs) offer a mechanism for putting side scan sonar transducers near potential targets together with telemetry, imaging, and oceanographic sensors, and can thus work together with low frequency sonar to develop holistic scatterscapes of oceanographic features, inclusive of information on species identity, orientation, behavior, abundance, individual size, and feature stability.

10:40

1aAO10. Scattering and reverberation from fish schools in the 500–1500 Hz band. Arthur Newhall, Ying-Tsong Lin, James F. Lynch (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 210 Bigelow Lab, MS #11, Woods Hole, MA 02543, anewhall@whoi.edu), Glen G. Gawarkiewicz (Physical Oceanogr., Woods Hole Oceanogr., Woods Hole, MA), and Thomas M. Grothues (Rutgers Univ., Tuckerton, NJ)

We will report on the results from an experiment off Cape Hatteras, North Carolina, to look at scattering and reverberation from fish schools in the 500–1500 Hz band. The experiment, which was performed during the period May 12–29, 2012, was a joint acoustics, biology, and physical oceanography effort, with distinct, but coordinated, goals in each area. Acoustically, we wished to examine the scattering of sound from fish schools over a full range of azimuthal angles. To do this, we employed a source mounted on an autonomous vehicle and a moored, four element array receiver. The source traveled around the fish school and the receiver, giving the desired angular diversity. Video images, sidescan sonar, and direct sampling of the school allowed us to quantify the *in-situ* scattering field. Estimates for attenuation and scattering versus azimuthal angle will be presented. Directions for analysis and further research will be discussed.

10:55

1aAO11. Deep sea organisms density estimation using a Dual Frequency Identification Sonar (DIDSON). Giacomo Giorli (Oceanogr., Univ. of Hawaii at Manoa, 1000 Pope Rd., Honolulu, HI 96822, giacomog@hawaii.edu), Adrienne M. Copeland, Whitlow W. Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii at Manoa, Kaneohe, HI), and Jeff Polovina (Pacific Island Fisheries Sci. Ctr., NOAA, Honolulu, HI)

Estimating the density of organism living in deep sea scatting layers is of key importance for understanding the biomass in the mesopelagic layers. Scientific echosounders are routinely used for this task; however, new imaging sonar technologies pose the opportunity for estimating density of organism, as well as identification at smaller scales. During the 2013 NOAA KONA Integrated Ecosystem, a Dual Frequency Identification Sonar (DIDSON) (SoundMetrics Inc.) was used to estimate the density, length of organisms in the deep sea scattering layers during nighttime and daytime along the KONA coast of the island of Hawaii. At each station, an EK60 38 kHz echosounder was used to find the depth of the deep sea scattering layers, and the DIDSON was lower into the layer (or layers if two were present) (about 500 and 600 m), and underneath the deeper layer (about 800 m). A total of 4621 organisms were counted and sized. We estimated densities ranging from 6 to 1 organism/m^3. Density shows some variation between locations and depth and organism as big as 3 m were sighted.

11:10

1aAO12. Bioacoustic absorption spectroscopy measurements at the shelf beak off Oregon. Orest Diachok (Appl. Phys. Lab., Johns Hopkins Univ., 11100 Johns Hopkins Rd., Laurel, MD 20723, orestdia@aol.com)

This paper describes the results of a Bioacoustic Absorption Spectroscopy (BAS) experiment, which was conducted at a biological hot spot at the shelf break off Oregon in August 2012. The location of the hot spot was identified by a NW Fisheries Science Center survey. This experiment included coincident measurements of transmission loss (TL), fish layer depths, fish length distributions, and continuous temperature profiles. The objective was to derive the bioacoustic parameters of hake (a physoclist), the dominant species in this region and other species from measurements of TL vs. range (0–10 km), frequency (0.3–5 kHz), depth (0–260 m), and time of day, and demonstrate consistency with trawl and echo sounder data. TL measurements were conducted between a moving, ship-deployed, broadband source and a 24 element, water column-spanning vertical array, which were provided by the Naval Research Laboratory. This was the first BAS experiment that targeted a physoclist. Previous BAS experiments targeted physostomes. The TL data exhibited absorption lines that were as high as 2 dB/km. Absorption lines at relatively high frequencies, which were observed on near-surface and mid-ocean hydrophones, are attributed to myctophids. Absorption lines, which were observed at relatively low frequencies on near-bottom hydrophones, are attributed to hake.

11:25–11:50 Panel Discussion

Session 1aBA

Biomedical Acoustics: Breast Ultrasound I

Koen W. A. van Dongen, Cochair
Lab. of Acoust. Wavefield Imaging, Faculty of Appl. Sci., Delft Univ. of Technol., P.O. Box 5046, Delft 2600 GA, Netherlands

Timothy E. Doyle, Cochair
Physics, Utah Valley Univ., MS 179, 800 W. University Parkway, Orem, UT 84058-5999

Chair's Introduction—7:55

Invited Papers

8:00

1aBA1. Three-dimensional ultrasound computer tomography at Karlsruhe Institute of Technology (KIT). Nicole V. Ruiter, Michael Zapf, Torsten Hopp, Ernst Kretzek, and Hartmut Gemmeke (Inst. for Data Processing and Electronics, Karlsruhe Inst. of Technol., Postfach 3640, Karlsruhe 76021, Germany, nicole.ruiter@kit.edu)

The KIT 3D USCT surrounds the breast with ultrasound transducers on a 3D aperture and emits and receives nearly spherical wave fronts for synthetic aperture focusing. This full 3D system achieves isotropic 3D resolution, has a nearly spatial invariant point spread function, and allows fast data acquisition. The 3D USCT device is equipped with 2041 ultrasound transducers. The acquisition is carried out by sequentially selecting a single emitter, sending a chirp at 2.5 MHz center frequency and recording the transmitted and reflected waves with all receivers. Rotational and translational movement of the aperture is applied to enhance the image contrast. Up to 40 GB of raw data is acquired with 480 parallel channels for digitization at 12 bit and 20 MHz sampling frequency. In a first pilot study, ten patients with different lesions were imaged. Speed of sound, attenuation, and reflection images of each patient were derived from the raw data. Overlaid volumes of the modalities show qualitative and quantitative information at a glance. The results are promising because the breasts' tissue structures and cancerous lesions could be identified in the USCT images.

8:20

1aBA2. Quantitative three dimensional nonlinear inverse scattering and reflection breast imaging: Initial clinical results. James Wiskin, David Borup, Elaine Iuanow, John Klock, and Mark Lenox (CVUS, LLC, Inc., 3216 Highland Dr., Ste. 100, Salt Lake City, UT 84106, jwiskin.cvus@gmail.com)

Water bath breast scanners utilize either ray based or inverse scattering techniques for the quantitative images. The 3D inverse scattering approach we use requires a 3D forward and back-propagation problem to be solved, resulting in ~1.2 mm resolution images. The resulting speed of sound map is then used in a 3D inhomogeneous, eikonal equation based reflection algorithm to account for refraction effects, and yield a 3D co-registered speckle free 360 degree compounded B-scan like volume. There is no harmful ionizing radiation, no compression, and no required contrast agents for proper utilization of our device. The quantitative transmission ultrasound (QTUS) images are independent of operator skill. The patient lies prone on a full length table with the breast pendant in the water bath. The breast is gently immobilized with the use of a breast retention pad that magnetically "attaches" to a magnetic retention rod for imaging of the breast. Scan time for each breast is approximately 10 min. We will show full 3D quantitative images obtained from consented, anonymous patients from several academic collaborating institutions. We will compare our QTUS images with mammographic, MRI, and hand held US images, and correlate biopsy results where appropriate.

8:40

1aBA3. Clinical breast imaging with ultrasound tomography: A description of the SoftVue system. Neb Duric, Peter Littrup (Oncology, Karmanos Cancer Inst., 4100 John R, Detroit, MI 48201, duric@karmanos.org), Olivier Roy, Cuiping Li, Steve Schmidt, Xiaoyang Cheng, and Roman Janer (Delphinus Medical Technologies, Plymouth, MI)

We describe the technical design and performance of SoftVue, a breast imaging device based on the principles of ultrasound tomography. SoftVue's imaging sensor is a ring shaped transducer operating at 3 MHz and consisting of 2048 elements. Data acquisition is achieved through 512 receive channels. The transducer encircles the breast, which is immersed in warm water while the patient lies in a prone position. The transducer is translated vertically to acquire data from the entire breast. The acquired data are used to reconstruct images using tomographic inversions. The reconstruction engine is based around a blade-server design that houses multiple CPUs and GPUs. Separate algorithms are used to reconstruct reflection, sound speed, and attenuation images. A patient scan generates a stack of each type of image. The system was designed with the clinical goal of detecting and characterizing breast masses based on their biomechanical and morphological properties. Ongoing clinical studies are being used to assess the performance of the system under realistic clinical settings. Results of the clinical assessment are presented. [The authors acknowledge research support from the National Cancer Institute (5 R44 CA165320-03). Neb Duric and Peter Littrup also acknowledge that they have financial interests in the SoftVue technology.]

9:00

1aBA4. High definition ultrasound of the breast. Robert Kruger, Richard Lam, Daniel Reinecke, and Stephen Del Rio (OptoSonics, Inc., 108 Straight Rd., Oriental, NC 28571, bobkruger@optosonics.com)

We propose to develop high definition ultrasound (HD-US) based on a hemispherical synthetic aperture and backscattered sonic waves. This configuration will produce direct three-dimensional maps of ultrasound reflectivity of breast tissues and will display higher signal-to-noise, reduced speckle, and greater spatial resolution compared to ultrasound arrays based on linear or curved planar sampling apertures. Our prototype hemispherical array consists of four interdigitated sub-arrays, each having 128 discrete elements. Currently, this array is used to capture 3D photoacoustic images of the whole breast using a spiral scanning strategy to increase the field of view sufficiently to perform whole-breast screening—photoacoustic mammography (PAM). We will add a multiplexed pulser, pulse sequencer, and T/R switches to capture ultrasound reflectivity data sufficient to form a 3D ultrasound image using the same reconstruction strategy used in PAM. Full 3D image acquisition will take 1.7 min, the same as currently used for PAM data collection. HD-US will be combined with PAM to produce 3D images of soft tissue and microcalcifications. The 3D images will be co-registered with the PAM images of hemoglobin distribution in the breast using a single hemispherical transducer array.

9:20

1aBA5. Low-contrast lesion detection enhancement using pulse compression technique. Serge Mensah, Julien Rouyer, Arnaud Ritou, and Philippe Lasaygues (Acoust. and Mech. Lab., National Ctr. for Sci. Res., 31 Chemin Joseph Aiguier, Marseille 13402, France, mensah@lma.cnrs-mrs.fr)

Pulse compression methods greatly improve the quality of medical images. In comparison with standard broadband pulse techniques, these methods enhance the contrast-to-noise ratio and increase the probing depth without any perceptible loss of spatial resolution. The Golay compression technique is analyzed here in the context of ultrasonic computed tomography, first on a one-dimensional target and second, on a very low-contrast phantom probed using a half-ring array tomograph. The imaging performances were assessed based on both the point spread function properties and the image contrast-to-noise ratio. The improvement obtained in the image contrast-to-noise ratio (up to 40%) depends, however, on the number of coherently associated diffraction projections. Beyond a certain number, few advantages were observed. Advances in ultrasound computed tomography suggest that pulse compression methods should provide a useful means of optimizing the trade-off between the image quality and the probing sampling density. It could also be used to accelerate the reconstruction process during the examination of patients. The results of this study also suggest that when it is proposed to search for very low contrast lesions (such as diffuse lobar carcinomas in breast cancer), once the number of projections has been set for a given tomographic set-up, the image contrast, i.e., the probability of detection, can been enhanced by using low-power, high-energy Golay sequences

9:40–10:00 Break

Contributed Papers

10:00

1aBA6. The effect of crosstalk in a circular transducer array on ultrasound transmission tomography of breast. Krzysztof J. Opielinski, Piotr Pruchnicki (Faculty of Electronics, Wroclaw Univ. of Technol., Wyb. Wyspianskiego 27, Wroclaw 50-370, Poland, krzysztof.opielinski@pwr.wroc.pl), Wlodzimierz Roguski (DRAMINSKI Medical Instruments, Olsztyn, Poland), Mateusz Celmer, Tadeusz Gudra (Faculty of Electronics, Wroclaw Univ. of Technol., Wroclaw, Poland), Jaroslaw Majewski (Faculty of Telecommunications, Comput. Sci. and Elec. Eng., Univ. of Technol. and Life Sci. in Bydgoszcz, Bydgoszcz, Poland), Mariusz Bulkowski, Tomasz Piotrowski, and Andrzej Wiktorowicz (DRAMINSKI Medical Instruments, Olsztyn, Poland)

Various types of undesirable crosstalk can occur in ultrasonic arrays consisting of a number of elementary piezoelectric transducers. Crosstalk is a result of deficiency in electrical or mechanical isolation between array elements. It is a function of proximity between piezoelectric transducers of the array or between connections. Small piezoceramic transducers tightly located inside of a circle (cylinder or sphere in 3-D systems) are mostly used for *in vivo* ultrasound tomography imaging of breast tissue. This means that proper switching of transducers makes it possible to quickly acquire tomographic transmission and reflection data for all directions around the object. A problem with such arrangements is the occurrence of crosstalk, which introduce specific errors to measurement data. Transmission and reflection tomographic images reconstructed based on ultrasonic measurements with crosstalk are distorted in a characteristic way. This work analyzes the effect of crosstalk in a circular transducer array on ultrasound transmission tomography imaging—especially, when used for examination of breast tissue.

10:15

1aBA7. Improving ultrasound-based estimates of vector displacement fields in elastography applications. Sanjay S. Yengul, Olalekan A. Babaniyi (Mech. Eng., Boston Univ., 69 Spyglass Hill Dr., Ashland, MA 01721, ysanjay@gmail.com), Faik C. Meral, Bruno Madore (Radiology, Brigham and Women's Hospital, Boston, MA), and Paul E. Barbone (Mech. Eng., Boston Univ., Boston, MA)

Ultrasound elastography is based on estimating tissue displacement using ultrasound imaging technology. The displacement field of interest may be the result of either an external quasistatic deformation (i.e., strain imaging), or a propagating shear wave (i.e., radiation force imaging), or some other excitation. Ultrasound image data with standard beamforming can provide very precise measurements of soft tissue displacement in the axial direction, i.e., in the direction of the ultrasound beam. Lateral (and elevational) displacement estimates are relatively noisy. This observation demonstrates that the quality of displacement estimates depends on the ultrasound image formation method. In this contribution, we evaluate the precision of axial and lateral displacement estimates obtained using different image formation methods. Ultrasound data from a Verasonics V-1 system (Redmond, WA) is used, which provides access to the raw signal from each element of the transducer array, thereby allowing beamforming to be done in software as a post-processing step. Using information from a mechanics-based model [Babaniyi *et al*., J. Acoust. Soc. Am. **134**, 4011 (2013)] further improves the precision of the displacement field estimate. We will present the results and discuss the next steps.

10:30

1aBA8. Boundary conditions in quantitative elastic modulus imaging.
Daniel T. Seidl, Olalekan A. Babaniyi (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, tseidl@bu.edu), Assad A. Oberai (Mech., Aerosp. and Nuclear Eng., Rensselaer Polytechnic Inst., Troy, NY), and Paul E. Barbone (Mech. Eng., Boston Univ., Boston, MA)

Quantitative elastic modulus imaging from quasistatic strain fields has several advantages over other approaches. It requires no specialized hardware, provides spatial resolution nearly commensurate with Bmode, and may be extended to quantitative nonlinear modulus imaging in a straightforward way. It has the drawback, however, of requiring the solution of a complex inverse problem with an ultrasound measured displacement field. The most common approach to solving the inverse problem is iterative optimization. To solve the inverse problem, the experimental configuration is simulated. Material property distributions in the simulated experiment are then varied until the simulated deformation field matches the observed deformation field. The weak link in this process is uncertainty in the experimental conditions. In the context of iterative inversion, this translates into uncertainty in the boundary conditions of the forward model. In this talk, we discuss how different choices of displacement and/or traction boundary conditions affect the inverse problem's solution using phantom and clinical breast data. We show that a Bayesian estimate of the displacement field in the face of uncertain boundary conditions can be implemented by spring finite elements on the domain boundary. [Authors gratefully acknowledge funding from NSF and NIH (NSF SI2 Grant No. 1148111; NIH NCI-R01CA140271).]

10:45

1aBA9. Boundary condition-free elastic modulus reconstructions from ultrasound measured quasi-static displacements.
Olalekan A. Babaniyi, Daniel T. Seidl (Mech. Eng., Boston Univ., 730 Commonwealth Ave., Boston, MA 02215, lekanb@bu.edu), Assad A. Oberai (Mech., Aerosp., and Nuclear Eng., Rensselear Polytechnic Inst., Troy, NY), Michael S. Richards (Dept. of Surgery, Univ. of Rochester, Rochester, NY), and Paul E. Barbone (Mech. Eng., Boston Univ., Boston, MA)

Quantitative elastic modulus imaging from quasi-static displacement data requires the solution of an inverse elasticity problem. The inverse problem formulation generally requires specification of either displacement or traction boundary conditions. Most current ultrasound devices are not capable of measuring traction data, and the measured displacement field is noisy. The incomplete and imprecise nature of the available boundary information often necessitates that educated guesses be made in order to have adequate knowledge of the boundary conditions to compute mechanical properties. These assumed boundary conditions, however, can to lead to errors in the reconstructions. This abstract proposes a method to perform reconstructions without knowing the boundary conditions *a priori*. This method relies on using the constrain imposed by the equilibrium equation and an optimization algorithm to estimate the modulus field. This method was verified with simulated displacement data, validated with phantom displacement data, and applied to *in-vivo* displacement data measured from patients with breast masses. [Authors gratefully acknowledge funding from NSF and NIH (NSF Grant No. 50201109; NIH NCI-R01CA140271).]

11:00

1aBA10. Comparative analysis of small versus large transducers for high-frequency ultrasonic testing of breast cancer.
Madison J. Peterson, Nathan M. Bliss (Biology, Utah Valley Univ., 904 N 960 W, Orem, UT 84057, madisonpr@aol.com), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

High-frequency ultrasound (20–80 MHz) has been found to be sensitive to margin pathology from breast cancer surgery. In order to improve the resolution and sensitivity of this method, transducers are needed that have smaller piezoelectric elements than those currently in use. This study's purpose was to determine if small-element transducers (Blatek, 50 MHz, diameter $<$2 mm) produce similar results as those from large-element transducers (Olympus NDT, 50 MHz, 6.35-mm diameter). Pulse-echo and through-transmission measurements were performed on bovine heart tissue and 10 phantom specimens containing chopped nylon fibers and polyethylene microspheres. The density of peaks in the ultrasonic spectra of the small or mini transducers (MT) paralleled those of the large transducers (LT) in the bovine tissue, with higher peak densities associated with connective tissue and lower peak densities with muscle tissue. The MT data from the phantoms showed greater variance than the LT data, indicating that the MT were more sensitive to the heterogeneous wavefields arising from microsphere scattering. Additional *in vivo* testing is currently being performed on breast tumors grown in mice treated with Avastin. Small-element transducers may ultimately provide *in vivo* cancer detection in margins, allowing more precise excision of cancerous tissue and thus eliminating follow-up surgeries.

Session 1aED

Education in Acoustics and Physical Acoustics: Tools for Teaching Advanced Acoustics

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

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

Invited Papers

8:30

1aED1. An information-rich learning environment for instruction in acoustics, part 3. Robert Celmer (Acoust. Prog. & Lab, Univ. of Hartford, 200 Bloomfield Ave., West Hartford, CT 06117, celmer@hartford.edu)

The static written word has always had its limits when it comes to learning about sound. Much of the subject matter is dynamic, multifaceted, and of course, aural. This presentation will describe additional multimedia materials developed for in-class presentation and self-paced review exercises for acoustics instruction at the University of Hartford. Some of the materials were developed using certain authoring applications, draw and animation programs, sound manipulation software, as well as 3D-CAD and spectral analysis applets. Audio equipment used in class as well as acoustic treatments of the classroom/listening environment will be described. This approach to acoustic pedagogy will be discussed in the context of a student-centered learning environment. New and updated demonstrations of the materials for the instruction of acoustical concepts as well as case studies will be presented.

8:50

1aED2. Fun with levitators. R. Glynn Holt (Mech. Eng., Boston Univ., Dept. of Mech. Eng., 110 Cummington Mall, Boston, MA 02215, rgholt@bu.edu)

The acoustic radiation force is enjoying something of a comeback on the celebrity circuit these days. Current applications are seen, for example, in elasticity imaging in biomedical ultrasound, and separation technologies in the biomedical and petroleum arenas. But of course the application of radiation force to acoustic levitation has a long history, and in this talk, we will explore levitation to illustrate the principles of radiation force in standing waves with sample inclusions. With a little luck, both large bubbles in water and liquid drops in air will be demonstrated.

9:10

1aED3. Animations illustrating the reflection of longitudinal sound wave pulses. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu)

The reflection of transverse wave pulses from a fixed or free boundary may be demonstrated physically in the classroom using an elastic string under tension. Animations of transverse wave pulses reflecting from hard and soft boundaries clearly indicate the phase relationship between incident and reflected signals. However, demonstrating similar phenomena for longitudinal waves difficult. The reversal of longitudinal waves in a spring may be physically demonstrated, but not the phase relationship between incident and reflected wave pulses. A microphone and digital oscilloscope may be used to demonstrate the phase change in pressure for reflections from the open end of a pipe, and the lack of phase change for reflections from a rigid end, but such demonstrations do not illustrate the longitudinal behavior of the wave motion at the boundary during reflection. This paper will showcase animations of longitudinal wave pulses reflecting from fixed and free boundaries, with an emphasis the appropriate phase changes upon reflection, with clarification of relationships between particle displacement, particle velocity, and pressure for longitudinal waves traveling in positive and negative directions. Discussion will include ways these animations may be used to improve student understanding.

9:30

1aED4. Bringing MATLAB into the acoustics class. Joseph F. Vignola, Aldo A. Glean (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20064, vignola@cua.edu), Teresa J. Ryan (Eng., East Carolina Univ., Greanville, NC), and Diego Turo (BioEng., George Mason Univ., Fairfax, VA)

The Catholic University of America has a history of graduate education in acoustics that ranges back to the 1930s. Many of the students served by CUA have been and continue to be working professionals in careers related to acoustics. Over time, we have observed that many of our students come to us having developed an impressive depth of knowledge in their specialty. However, some of these students have little of the formal acoustics training needed to provide broader context. This presentation will discuss teaching practices designed to serve this cohort as well as more traditional graduate students. We create an experiential learning environment that capitalizes on the fact that many simple but powerfully instructive measurements can be made with now ubiquitous items. All modern laptops

have both speakers and a microphone. A laptop, coupled with a library of MATLAB code, gives students the opportunity to explore many important topics. This presentation includes examples from room acoustics, musical acoustics, and elastic wave propagation.

9:50

1aED5. Teaching acoustics in the time domain and frequency domain: Going back and forth. Karim G. Sabra (Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr., NW, Atlanta, GA 30332-0405, karim.sabra@me.gatech.edu)

Our common experience with acoustic signals typically occurs in the time domain: for instance when listening to a conversation or recording an audio signal. On the other hand, acoustics in the classroom is often taught in the frequency domain; for instance, doing so can offer clear simplification of the wave equation (e.g., when solving the Helmholtz equation). Exact time-domain solutions can ultimately be obtained back from these frequency solutions using Fourier synthesis; but it typically requires numerical methods or extra analytical work. However, based on classroom experiences, it can be valuable that students develop an intuitive interpretation of these frequency-domain solutions in the time-domain without using any computers or actual inverse Fourier transform evaluation. For instance, a question one may ask students is "how would you predict the overall shape and features of time-domain waveform corresponding to this specific frequency-domain solutions we just derived, i.e., as if you were to perform an actual experiment and just recorded a time-domain waveform." I will discuss various basic examples covered in the classroom such as modal propagation and dispersion effects in a waveguide.

10:10–10:20 Break

10:20

1aED6. Using Python to teach mathematics, physics, and acoustics. Derek C. Thomas and Benjamin Y. Christensen (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, dthomas@byu.edu)

Advanced technical courses often suffer from a lack of interactive materials. Common tools to remedy this deficiency include MATLAB and MATHEMATICA, both of which can be prohibitively expensive to obtain outside of the university environment. Python is a scripted language that is easy to read and use and is rapidly emerging as a lingua franca for scientific computing due to the flexibility and facility of the language, the large and active community, and the large number of high quality scientific libraries that are available in Python. Python provides a free and open source tool to develop classroom materials that students can modify and extend. We discuss the use of Python in teaching advanced topics in mathematics, physics, and acoustics. Examples are drawn from courses in acoustics, mechanics, and mathematical and computational physics

10:40

1aED7. Resources for teaching near-field acoustical holography in advanced acoustics courses. Kent L. Gee, Tracianne B. Neilsen, and Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu)

The principles of planar near-field acoustical holography (NAH) can be used to motivate discussion of, or illustrate, physics of sound radiation or topics in signal processing. These include separable geometries, the Helmholtz equation, spatial Fourier transforms and the wavenumber domain, superposition of waves, the relationship between pressure and particle velocity, near and far fields, the radiation circle and evanescence, filtering, and signal-to-noise ratio. This paper describes the incorporation of NAH as part of a graduate-course unit on structural acoustics. Resources discussed, and which will be made available to educators, include a basic NAH processing script and example data collected by students using an automated positioning system in Brigham Young University's fully anechoic chamber.

Contributed Papers

11:00

1aED8. Software usage for synthesized sound in acoustics education. Jennifer K. Whiting, Katherine H. Fortney, Tracianne B. Neilsen, and Kent L. Gee (Dept. of Phys. and Astronomy, Brigham Young Univ., C110 ESC, Provo, UT 84606, lundjenny@comcast.net)

Musical acoustics engages students of many backgrounds when studying acoustics. Free software in the form of digital audio workstations can be used to provide students with a hands-on instruction to concepts of synthesis. Students can easily manipulate variables to add or detract from the realism of synthesized sound. This process of synthesizing sound provides a means for students to understand the concepts of additive and subtractive synthesis, attack and decay times, reverberation times, and filters. Application of free digital audio workstation software in an undergraduate class at Brigham Young University and in the recently revamped ASA Outreach workshop will be discussed.

11:15

1aED9. Architectural acoustics illustrated, animated, designed, and built. Michael Ermann, Nawazish Nanji (Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu), Vinny Argentina, Matt Yourshaw (School of the Visual Arts, Virginia Tech, Blacksburg, VA), Marie Zawistowski, Keith Zawistowski, Lauren Duda, Megumi Ezure, Samantha Stephenson, Taylor Terrill, Ian Shelton, Samantha Yeh, Kyle Lee, Huy Duong, Brent Sikora, Leo Naegele, Tyler Atkins, Derek Ellison, Margaret Nelson, Leah Schafer, and Emarie Skelton (Architecture + Design, Virginia Tech, Blacksburg, VA)

In this line of pedagogy, architectural acoustics is filtered through the graphic and built language of architecture. First, a series of animations were created to explain room acoustics to architecture students. The impulse response as a concept is inherently spatial and dynamic. It can be explained with text, and it can be explained more clearly when illustration is included,

but because the path of sound and the loudness at a receiver fluctuates with time, it can be best explained with a narrated animation (available online). Second, the physics of sound, room acoustics, and noise control were illustrated as part of a book, *Architectural Acoustics Illustrated* (Wiley, 2015). Building material choices, spatial relationships, best-practices, and data were explored through drawing; non-obvious and counter-intuitive graphic findings are presented. Finally, 15 architecture students explored room acoustics through ray tracing and auralization software. Then they designed and built an amphitheater for the town of Clifton Forge, Virginia. The completed project was widely published, was the subject of a documentary film, and earned award recognition from the American Institute of Architects.

11:30

1aED10. Demonstrations and laboratory experiments in a senior level acoustics course. Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

The Physics Department at the U.S. Naval Academy has a senior level four credit course (SP436) in acoustics (called "Acoustics") that features a well-equipped laboratory including an anechoic chamber. The course is populated by Physics Majors along with a few Engineering Majors at times. This presentation will show how MATHEMATICA 9 is used in the laboratory portion of the course to enhance lecture topics and student computational assignments. It is "hands-on" which helps motivate and enhance learning. Students learn the fundamentals of Mathematica as part of the laboratory experience. Tasks after data collection, including analysis and write-up, are done with this software. Workstations include laptop personal computers, spectrum analyzers, oscilloscopes, laser Doppler vibrometers, and various accelerometers, mics, hydrophones, and ultrasonic transducers. A demonstration of standing waves in a cylindrical cavity will be presented as a representation of some of our featured lab experiments—including: Helmholtz resonators, linear and nonlinear vibration of a circular membrane, or circular elastic plate, flexural waves on a thin bar, Chladni plates, the hanging oscillating chain, spectral analysis (Fourier series, Fourier integral), sound speed vs. temperature and salinity, acoustic landmine detection, moving coil loud speaker, waves on strings, transmitting arrays, and wave guide studies.

11:45

1aED11. YouTube EDU: Inspiring interest in acoustics through online video. Michael B. Wilson (Acoust., Penn State, 1649 Highlandon Ct, State College, PA 16801, wilsomb@gmail.com)

The Internet is changing the face of education in the world today. More people have access to more information than ever before. The Khan Academy, iTunes U, YouTube EDU, and other programs are providing educational content for free to millions of Internet users worldwide. This content ranges from interesting facts that introduce a topic to entire undergraduate courses. And just as acoustics is an underrepresented science at the secondary and undergraduate levels, acoustics is an underrepresented science in the world of online education. But that is changing. Online content is being created focusing on clarifying misconceptions and sparking an interest in the field of acoustics. These videos are available without charge to everyone in the world.

MONDAY MORNING, 5 MAY 2014 BALLROOM A, 8:00 A.M. TO 12:00 NOON

Session 1aPP

Psychological and Physiological Acoustics: Temporal Processing; Pitch and Timbre Perception; Speech Perception (Poster Session)

Agnès C. Léger, Chair
Res. Lab. of Electronics, Massachusetts Inst. of Technol., Rm. 36-757, 77 Massachusetts Ave., Cambridge, MA 02139

All posters will be on display from 8:00 a.m. to 12:00 noon, To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10.00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

1aPP1. Does fundamental frequency-segregation interact with dip listening or spatial unmasking? Thibaud Leclére, Mathieu Lavandier (LGCB, ENTPE-Université de Lyon, rue Maurice Audin, Vaulx-en-Velin, Rḩnes 69518, France, thibaud.leclere@entpe.fr), and Mickael L. Deroche (Dept. of Otolaryngol., Johns Hopkins Univ. School of Medicine, Baltimore, MD)

Differences in fundamental frequency (F0) and location between a speech target and a masker as well as amplitude modulations in the masker are helpful cues to improve speech intelligibility in cocktail party situations. Each cue has been thoroughly investigated independently in many studies, but it remains unclear whether they interact with each other. Experiment 1 examined potential interactions between F0-segregation and dip listening while experiment 2 examined interactions between F0-segregation and spatial unmasking. Speech reception thresholds were measured for a monotonized or an intonated voice against eight types of harmonic complex interferers. In experiment 1, the eight interferers varied in F0 contour (monotonized or intonated), mean F0 (0 or 3 semitones above the target) and broadband temporal envelope (stationary or 1-voice modulated). In experiment 2, interferers varied in F0 contour, mean F0 and spatial location (colocated or separated from the target). Thirty-two listeners participated in each experiment. The results will be presented and discussed.

1aPP2. Speech intelligibility and masking release using temporal fine structure and recovered envelope cues for normal-hearing and hearing-impaired listeners. Agnes C. Leger, Charlotte M. Reed, Joseph G. Desloge, Jayaganesh Swaminathan, and Louis D. Braida (Res. Lab. of Electronics, Massachusetts Inst. of Technol., Rm. 36-757, 77 Massachusetts Ave., Cambridge, MA 02139, aleger@mit.edu)

Consonant-identification ability was examined in normal-hearing (NH) and hearing-impaired (HI) listeners in the presence of continuous and 10-Hz square-wave interrupted speech-shaped noise. The speech stimuli (16 consonants in a-C-a syllables) were processed to present envelope (ENV) cues,

temporal fine-structure cues (TFS), or envelope cues recovered from TFS speech (RENV). ENV- and TFS-speech was generated by extracting the ENV or TFS component of the unprocessed speech in N adjacent bands (N = 40 for ENV-speech, N = 1 or 4 for TFS-speech; range = 80–8020 Hz). RENV-speech was generated by extracting the ENV component of both types of TFS-speech in 40 adjacent bands. NH listeners were tested at an SNR of −-10 dB and individual HI listeners were tested with SNR in the range of −-6 to + 5 dB. These values of SNR yielded consonant-identification scores of roughly 50%-correct for intact speech in continuous noise for each listener. HI listeners had poorer speech scores than NH listeners. For both groups, scores with TFS- and RENV-speech were very similar. Scores were higher in interrupted noise than in continuous noise (indicating substantial release from masking), except for unprocessed- and ENV-speech for HI listeners. Audibility, frequency selectivity, and forward masking were estimated for each listener and compared with speech identification.

1aPP3. Aging and lexical neighborhood effects in competing speech perception. Karen S. Helfer, Angela Costanzi, and Sarah Laakso (Commun. Disord., Univ. of Massachusetts Amherst, 358 N. Pleasant St., Amherst, MA 01002, khelfer@comdis.umass.edu)

Little is known about the extent to which lexical neighborhood effects are influenced by the presence of to-be-ignored messages. When the competing signal is understandable speech, words in the masker may activate their own lexical neighbors, causing increased competition for word identification. Older adults may have particular difficulty inhibiting lexical activation. This poster will present results of an examination of lexical neighborhood influences on competing speech perception. Sentences were developed in which we manipulated the neighborhood density and frequency of key words to create lexically easy (low density or high frequency of usage) and lexically difficult (high density or low frequency) stimuli lists. Pairs of sentences were created in which one sentence was lexically easy (in terms of density or frequency) and one was difficult. The target sentence was always spoken by the same talker and could be either lexically easy or difficult. Participants were younger (18–23 years), middle-aged (45–59 years), and older (>60 years) adults; they also completed a battery of cognitive tests. This poster will show results of analyses comparing lexical neighborhood effects among the participant groups, as well as how these effects are influenced by hearing loss and cognitive abilities. [Work supported by NIH DC012057.]

1aPP4. Relationship between pitch and rhythm perception with tonal sequences. Sandra J. Guzman, Robert Almeida, Karson Glass, Cody Elston (Audio Arts & Acoust., Columbia College Chicago, 3734 Kenilworth, Berwyn, IL 60402, sguzman@colum.edu), Valeriy Shafiro, and Stanley Sheft (Commun. Disord. and Sci., Rush Univ. Medical Ctr., Chicago, IL)

Past work has shown varying degrees of relationship between pitch and rhythm perception. Current work investigated the relationship between pitch and rhythm processing in a four-tone sequence-reconstruction task which places additional demands on short-term memory. 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. When only one stimulus attribute was randomized, performance was significantly better with sequences defined by pitch rather than rhythmic variation. Combining pitch and rhythmic variations led to a slight improvement in performance, while the introduction of extraneous variation had little to no effect when either pitch or rhythm defined the target sequence. Overall, there was a wide performance range across listeners with listeners clustered primarily by ability to use pitch information. [Work supported by NIH.]

1aPP5. Examining the influence of forward, backward, and simultaneous notched noise on the mid-level hump in intensity discrimination. Elin Roverud and Elizabeth A. Strickland (Speech, Lang., and Hearing Sci., Purdue Univ., 712 Cincinnati St., Lafayette, IN 47901, eroverud@purdue.edu)

Psychoacoustical intensity discrimination limens (IDLs) for high frequency, short duration pedestals are poorer at mid levels than at lower and higher levels. It has been theorized that this so-called midlevel hump (MLH) may reflect mid-level cochlear compression, whereas the improvement at higher levels may be due to spread of excitation cues. To characterize intensity discrimination within a single frequency channel, researchers (e.g., Plack, 1998) have used simultaneous notched noise (NN) to limit cues from the spread of excitation to other channels. However, additional effects of the NN on the pedestal remain a matter of debate. The NN may provide a reference against which intensity judgments are made. Additionally, the NN may produce excitatory masking in the pedestal frequency channel, suppress the pedestal, and evoke cochlear gain reduction via the medial olivocochlear reflex. These latter two mechanisms change the basilar membrane compression slope, but operate over different time courses. In the present study, we examine the MLH with different durations of forward, backward, and simultaneous NN and pure tone maskers to isolate these potential mechanisms. Results will be interpreted using a computational model of the auditory system. [Research support provided by grants from NIH (NIDCD): R01-DC008327 and the Purdue Research Foundation.]

1aPP6. Influence of context on the relative pitch dominance of individual harmonics. Hedwig E. Gockel, Sami Alsindi, Charles Hardy, and Robert P. Carlyon (MRC-Cognition and Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB2 7EF, United Kingdom, hedwig.gockel@mrc-cbu.cam.ac.uk)

There is evidence that the contribution of a given harmonic in a complex tone to residue pitch is influenced by the accuracy with which the frequency of that harmonic is encoded. We investigated whether listeners adjust the weights assigned to individual harmonics based on acquired knowledge of the reliability of the frequency estimates of those harmonics. In a two-interval forced-choice task, seven listeners indicated which of two 12-harmonic complex tones had the higher overall pitch. In context trials (60% of all trials), the fundamental frequency (F0) was 200 Hz in one interval and 200+ΔF0 Hz in the other. In different blocks, either the third or the fourth harmonic, plus (always) the seventh, ninth, and 12th harmonics were replaced by narrowband noises that were identical in the two intervals. Feedback was provided. In test trials (40% of all trials), the fundamental frequency was 200+ΔF0/2 Hz in both intervals and either the third or the fourth harmonic was shifted slightly up or down in frequency. There were no narrowband noises. Feedback was not provided. The results showed that substitution of a harmonic by noise in context trials significantly reduced the contribution of that harmonic to pitch judgments in the test trials.

1aPP7. How is periodicity pitch encoded on basilar membrane of the cochlea? Takeshi Morimoto (Sensory and Cognit. Neural System Lab., Graduate School of Life and Medical Sci., Doshisha Univ., 8-16-10, Kanou, Higashiosaka-shi 578-0901, Japan, master1020scncl@gmail.com), Kohta I. Kobayasi, and Hiroshi Riquimaroux (Sensory and Cognit. Neural System Lab., Graduate School of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan)

Pitch is the perceptual correlate of periodicity of sound. Even if all the energy at the fundamental frequency (F0) is removed, a periodic complex tone retains the same pitch of missing fundamental frequency. The periodicity pitch has been mostly discussed in the central auditory system while less reported in the peripheral system. The purpose of this study was to investigate how the periodicity pitch is encoded in the cochlea. Cochlear microphonics (CM) was recorded from the round window for confirming frequency characteristics of vibration of the basilar membrane (BM) to periodic complex tones in order to generate F0. The experiments were carried out in the condition which a frequency component corresponding to F0 or frequency components of the sound stimuli were masked by low or high frequency band pass noises respectively. The forward masking did not have temporal overlap of the sound stimuli on the BM. The results showed that the frequency component of F0 existed in the CM. The frequency component of F0 decreased for high frequency band pass masking but little for low frequency band pass masking. The findings suggest that the periodicity pitch is encoded using rates of amplitude modulation rather than the location on the BM.

1aPP8. Pitch discrimination with harmonic and inharmonic tone complexes. Lars Bramsløw and Niels H. Pontoppidan (Eriksholm Res. Ctr., Rørtangvej 20, Snekkersten 3070, Denmark, lab@eriksholm.com)

The measurement of sensitivity to temporal fine structure (TFS) in listeners can be measured by using a linear frequency-shift of a harmonic tone complex. However, the linear frequency shift of the harmonic tone complex breaks the harmonic structure, introducing a harmonic-inharmonic cue in addition to the pitch shift. In the present study, we investigated the relative contributions of frequency shift and harmonicity in normal-hearing listeners, using harmonic-harmonic, harmonic-inharmonic, and inharmonic-inharmonic shifts for unresolved tone complexes. A two-down-one-up adaptive method was used to measure the frequency shift threshold. Our results show that both inharmonic variants of the frequency shift have lower detection thresholds than the harmonic-only shift. The two inharmonic conditions are not different, indicating that the linear shift threshold is not driven by a harmonic-inharmonic cue. The effect is the same for all frequencies tested here. The results are discussed in relation to both excitation pattern and temporal fine structure models.

1aPP9. Masked speech recognition in school-age children and adults: Effects of age, masker type, and context. Lori Leibold (Allied Health Sci., The Univ. of North Carolina at Chapel Hill, 3122 Bondurant Hall, CB#7190, Chapel Hill, NC 27599, leibold@med.unc.edu), Emily Buss, and Joseph W. Hall (Otolaryngology/Head and Neck Surgery, The Univ. of North Carolina at Chapel Hill, Chapel Hill, NC)

This study evaluated the influences of age, masker type, and context on masked speech recognition. A repeated-measures design compared the speech recognition thresholds of two age groups of children (5–7 and 9–13 years) and a group of adults (19–33 years) in a continuous speech-shaped noise or a two-talker speech masker. Target stimuli were disyllabic words that were familiar to young children. Masker level was fixed at 60 dB SPL, and signal level was adapted to estimate the SNR required for 70.7% correct performance. Each listener completed testing in each masker condition using both an open-set task requiring a verbal response, and a 4AFC closed-set task requiring a picture-pointing response. Consistent with previous studies, and regardless of response context, larger and more prolonged child-adult differences were observed in the two-talker compared to the speech-shaped noise masker. Poorer performance was observed for all three age groups using the open-set compared to the closed-set context. This performance gap was similar across the three age groups in the speech-shaped noise masker, but a developmental effect was observed in the two-talker masker. Specifically, the decrement in performance using the open-set compared to the closed-set procedure increased with age in the two-talker masker.

1aPP10. Gap detection in school-age children and adults: Effects of marker center frequency and ramp duration. Heather Porter, Emily Buss, Joseph W. Hall, and John H. Grose (Otolaryngol., Univ. of North Carolina at Chapel Hill, 170 Manning Dr., Chapel Hill, NC 27599-7070, heather_porter@med.unc.edu)

Data on the development of auditory temporal processing varies widely across studies, depending on the particular stimulus (e.g., marker center frequency) and the methods used to quantify performance. Recently, gap detection thresholds were observed to be adult-like later in childhood for narrowband low-fluctuation noise (with 40-ms ramps) than wideband Gaussian noise (with 4-ms ramps; Buss *et al.*, 2013). These results could reflect relatively protracted development of the ability to detect gaps in spectrally narrow stimuli compared to broader stimuli, irrespective of inherent stimulus fluctuation. That is, the use of across-channel cues that benefit gap detection for wideband stimuli could develop early in childhood. Alternatively, age effects in gap detection may depend on center frequency or ramp duration. The present study measured gap detection for low-fluctuation narrowband noise with either a low (500 Hz) or a high (5000 Hz) center frequency, and for wideband noise with short (4 ms) or long (40 ms) ramps. Listeners were normal-hearing 4- to 16-year-old children and adults. Results will be discussed in terms of the effect of center frequency, ramp duration, and across-frequency cues on gap detection for children and adults.

1aPP11. Priming in speech perception through captions and sign language. Richard L. Freyman, Gwyneth C. Rost, Derina S. Boothroyd, Charlotte Morse-Fortier, Amanda M. Griffin, and Sarah F. Poissant (Commun. Disord., Univ. of Massachusetts, 358 N. Pleasant St., Amherst, MA 01003, rlf@comdis.umass.edu)

When listeners know the content of the message they are about to hear, distorted or partially masked speech appears dramatically more intelligible compared to when the content is unknown. In the current research this priming phenomenon was investigated quantitatively using a same-different task where the prime and auditory message match on only 50% of trials. The first experiment was concerned with the effect of the timing of typed captions relative to the auditory message. For nonsense sentences subjected to four different types of distortion and masking, optimum performance was achieved when the initiation of the text preceded the acoustic speech signal by 800 ms. Performance was slightly poorer with simultaneous delivery and much poorer when the auditory signal preceded the caption. The second experiment investigated whether priming effects could be observed when the prime was delivered by sign language rather than captions. Preliminary results with normal-hearing signers indicate that some benefits in same-different performance can be triggered by sign language primes. Confirmation of this result and extension to listeners with hearing impairment who use sign and speech together in simultaneous communication (SimCom) may inform best practices for the use of SimCom in educational and other settings. [Work supported by NIDCD 01625 and The Hearing Health Foundation.]

1aPP12. The role of bilingualism and musicianship on performance in a masked speech detection task. Charlotte Morse-Fortier, Giang Pham, and Richard L. Freyman (Commun. Disord., Univ. of Massachusetts Amherst, 1147A North Pleasant St., Amherst, MA 01002, cmorsefo@comdis.umass.edu)

Listening to an individual talker in the presence of multiple speakers is one of the most difficult auditory situations listeners encounter. Past research has shown that musicians and bilinguals have more robust neural responses to a target in a competing speech background compared to non-musicians and monolinguals, perhaps due to enhanced processing of fundamental frequency. In this study, a simple speech detection paradigm was used to minimize the effect of language comprehension while studying behavioral performance on a speech-masking task. Four groups of listeners detected English target words in the presence of two-talker babble, where all speakers were female. The task was conducted in spatial and non-spatial configurations to address conditions with low and high informational masking, respectively. The four groups tested were musicians, non-musician Asian tonal bilinguals, non-musician Spanish bilinguals, and non-musician English monolinguals. Results show that all subject groups experienced a significant release from masking in the spatial condition relative to the non-spatial condition. Musicians outperformed non-musicians in the non-spatial condition (high informational masking), but not in the spatial condition where energetic masking was presumed to dominate. Differences between the two bilingual groups and the monolingual group will be discussed. [Work supported by NIDCD 01625.]

1aPP13. The effects of amplitude envelope and context on auditory duration perception. Lorraine Chuen (Dept. of Psych., Neurosci. and Behaviour, McMaster Inst. for Music and the Mind, Psych. Bldg. (PC), Rm. 102, McMaster Univ., 1280 Main St. West, Hamilton, ON L8S 4K1, Canada, chuenll@mcmaster.ca) and Michael Schutz (School of the Arts, McMaster Inst. for Music and the Mind, Hamilton, ON, Canada)

Here, we extend research indicating that a sound's amplitude envelope (changes in intensity over time) affects subjective duration (Grassi and Darwin, 2006). Specifically, we explore whether amplitude envelope affects the strategies underlying duration perception by comparing "flat" (abrupt onset, sustain, abrupt offset) and "percussive" (abrupt onset followed by exponential decay) envelopes. Participants performed a two alternative forced choice task: judging which of two tones sounded longer. Trials were divided into blocks organized by envelope: (a1) uniform flat-flat, (a2) uniform percussive-percussive, and (b) mixed (uniform, percussive-flat, and flat-percussive). This was designed to either permit (a) or prohibit (b) envelope-specific listening strategies. Block order was counterbalanced across participants. An analysis of the first block (a between-subjects comparison of uniform and mixed block performance) indicated that performance on flat-flat trials was not significantly different for the two block types, whereas percussive-percussive trial performance was significantly worse for the mixed block. This suggests that participants optimally employ an envelope-specific strategy for the uniform block (a) when able to predict envelope, and a generalized strategy in the mixed block (b) when unable. Interestingly, there was no performance advantage for the uniform block when presented second, suggesting that contextual order effects may affect auditory duration perception.

1aPP14. Real-time implementation of a polyphonic pitch perception model. Nikhil Deshpande and Jonas Braasch (Dept. of Architecture, Rensselaer Polytechnic Inst., 220 3rd St., Troy, NY 12180, deshpn@rpi.edu)

This algorithm is a real-time implementation of a polyphonic pitch perception model previously described in Braasch *et al.* [POMA **19**, 015027 (2013)]. The model simulates the rate code for pitch detection by taking advantage of phase locking techniques. Complex input tones are processed through a filter bank, and the output of each filter is run through its own separate autocorrelation following the Licklider model. After conversion to polar form, analysis is done on the phase output of the autocorrelation, where the algorithm computes the time delay between localized peaks. This gives the fundamental period of the tone within a given filter; these values are then normalized by the magnitude output of the autocorrelation and combined with output data from the other filters to give full spectral information. The algorithm uses an adjustable running frame window to trade off between frequency resolution and rapid changes in pitch. The model can accurately extract missing or implied fundamental frequencies.

1aPP15. Infants ability to perceive changes in timbre. Bonnie K. Lau and Lynne A. Werner (Univ. of Washington, 523 Broadway East Unit 217, Seattle, WA 98102, bonniekwlau@gmail.com)

Recent studies have demonstrated that infants can ignore spectral changes in sequentially presented complex tones and categorize them on the basis of missing fundamental pitch. However, infants' ability to discriminate these spectral changes is unknown. This study investigated the ability of adults, 7- and 3-month-olds to perceive changes in the spectral centroid of harmonic complexes using an observer-based method. Stimuli were 500-ms complex tones, with 20-ms raised cosine onset/offset ramps presented at 70 dB SPL. All harmonics were generated up to 10 000 Hz then bandpass filtered with a -24 dB/octave slope around the center frequency (CF). The experiment consisted of five phases that presented complexes with a 15, 10, 5, 2, or 0.5% ΔCF. To demonstrate timbre discrimination, participants were required to ignore changes in the fundamental frequency of complexes which randomly varied between 170, 180, 190, 200, 210, and 220 Hz, and to respond only when the CF of the spectral centroid changed. Infants were tested on three conditions (15, 10, 5%, or 15, 2, 0.5%). Adult participants were tested on all phases until they reached a phase they could not discriminate. Preliminary results indicate that infant performance is comparable to adults, suggesting discrimination of timbre at 3 months.

1aPP16. Pitch shifts in scale alternated wavelet sequences and the prediction by auditory image model and spectral temporal receptive field. Minoru Tsuzaki (Faculty of Music, Kyoto City Univ. of Arts, 13-6 Kutsukake-cho, Oe, Nishikyo-ku, Kyoto 610-1197, Japan, minoru.tsuzaki@kcua.ac.jp), Toshio Irino (Faculty of Systems Eng., Wakayama Univ., Wakayama, Japan), Chihiro Takeshima (J. F. Oberlin Univ., Machida-shi, Japan), and Tohie Matsui (Tsukiba Adv. Res. Alliance, Univ. of Tsukiba, Tsukuba, Japan)

SAWS's are acoustic stimuli in which an impulse response of vocal tract and its scaled version are alternately placed in the time domain at a constant periodic rate. When the scale factor is close to unity (1.0), the perceived pitch corresponded to the original periodicity. As the difference in the scaling became large, the pitch tended to be matched to what corresponds to be lower than the original by an octave. One of the characteristics of this pitch shift was that the pitch chroma did not change. This sort of pitch continuum could not be realized by changing the fundamental frequency of harmonic complex tones, but could be realized by attenuating the odd harmonics of them. Two auditory models were used to predict this pitch shift phenomenon, i.e., AIM by Patterson's group; STRF by Shmma's group. Both models could predict the pitch shift by an octave, but AIM predicted the pitch ambiguity better than STRF. While it is easy to find the secondary local peak of periodicity besides the primary peak, the peak activity in STRF was singular in most cases. The results suggested that AIM could preserve the temporal fine structure better than STRF.

1aPP17. Discrete and continuous auditory feedback based on pitch and spatial lateralization for human-machine-interface control. Sylvain Favrot, Carolyn M. Michener, and Cara E. Stepp (Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, cmich@bu.edu)

The purpose of this study was to investigate auditory-motor learning via discrete and continuous auditory feedback using pitch and spatial lateralization. Sixteen subjects used facial surface electromyography (sEMG) to control a human-machine-interface (HMI). The output of the HMI was a lateralized harmonic tone. The fundamental frequency and lateralization (left-right ear) changed with the sum and the difference of the bilateral muscle signals. For eight participants, these changes were continuous, whereas the other eight participants received discrete feedback, in which the frequency of the tone was one of nine possible semitones (from midi note #64 to #75) and the lateralization was either left, center or right. Participants trained over three days. A mixed-models analysis of variance showed a significant effect of learning over sessions and a trend for increased performance for the group utilizing discrete feedback. Overall, information transfer rates using this purely auditory feedback averaged 38.5 bit/min by day 3, which is similar to results from similar systems utilizing visual feedback. These results show that with minimal training, auditory feedback can provide usable HMI control.

1aPP18. Speech, spatial, and qualities of hearing scale (SSQ): Normative data from young, normal-hearing listeners. Pavel Zahorik and Ann M. Rothpletz (Heuser Hearing Inst. and Div. of Communicative Disord., Dept. of Surgery, Univ. of Louisville School of Medicine, Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu)

The Speech, Spatial, and Qualities of Hearing Scale (SSQ) was developed to assess listeners' subjective sense of listening ability and listening experience in everyday complex situations that often involve spatial hearing. The SSQ is one of the very few instruments designed to measure such quantities and has been used extensively to assess functional hearing impairment and benefit resulting from hearing remediation strategy. Although a recent study examined the psychometric properties of the SSQ with a large sample of hearing-impaired listeners, little published data exist from normal-hearing listeners. The data that have been published suggest that even young normal-hearing listeners do not rate their subjective listening abilities at the most-proficient end of the scale in all the listening situations probed by the SSQ. The goal of this study was to examine this issue more fully, using a sample of 233 young (median age 21.2 years, 3.1-year IQR), normal-hearing listeners (pure tone thresholds $\leq$ 25 dB HL from 250—8000 Hz). Results provide normative data on each of the (self-administered) SSQ

items, and describe the psychometric properties of the SSQ for a young normal-hearing population. These data are intended to aid in the interpretation of SSQ results from other populations.

1aPP19. Ear effect and gender difference of spontaneous otoacoustic emissions in children with auditory processing disorder. Kimberly Zwissler (Univ. of Delaware, 1600 Rockland Rd., Wilmington, DE 19803, zwissler@udel.edu), Kyoko Nagao, L. Ashleigh Greenwood (Ctr. for Pediatric Auditory and Speech Sci., Nemours/Alfred I. duPont Hospital for Children, Wilmington, DE), Rebecca G. Gaffney (Univ. of Delaware, Wilmington, DE), R. Matthew Cardinale (College of Osteopathic Medicine, New York Inst. of Technol., Wilmington, Delaware), and Thierry Morlet (Ctr. for Pediatric Auditory and Speech Sci., Nemours/Alfred I. duPont Hospital for Children, Wilmington, DE)

Spontaneous otoacoustic emissions (SOAEs) are found in most healthy ears and can be used to measure the health of the cochlear structures and feedback mechanism. According to existing literature, right ears tend to exhibit greater numbers of SOAEs than left ears (Bilger *et al*., 1990) and females tend to show higher incidence of SOAEs than males (Moulin *et al*., 1993). The SOAE prevalence has not been extensively studied in children with auditory processing disorder (APD), a disorder with unknown etiology that reduces one's ability to process auditory information. This study examined the prevalence and ear advantage of SOAEs between genders in children diagnosed with APD. SOAEs were investigated in 19 children (7 girls and 12 boys) with APD and 24 typically developing children (14 girls and 10 boys) aged 7–12. Right ear advantage was more prevalent in control (71%) than APD subjects (42%). However, over 30% more females exhibited a right ear advantage than males in each group. Although the results are not significant, our findings indicate that the lack of right ear advantage for SOAE is more prevalent in children with APD, particularly in males, suggesting that cochlear mechanisms or their control might be somehow affected in APD.

1aPP20. Testing a nonlinear computational channel model for masker phase effects. Yonghee Oh, Evelyn M. Hoglund, and Lawrence L. Feth (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, oh.172@osu.edu)

The masked threshold differences produced by Schroeder-phase maskers are most often attributed to the non-linear response of the normal cochlea (Summers *et al*., 2003). The nonlinear properties of the basilar membrane (BM) cause different response to the positive and negative Schroeder-phase maskers (i.e., +SCHR and -SCHR maskers, respectively) based on signal level, temporal synchrony, and on- and off-frequency harmonic components (Kohlrausch and Sander, 1995, Carlyon and Datta, 1997a,b, and Summers, 2000). In this study, manipulation of harmonic components of the maskers was used to explore nonlinear aspects of BM motion produced by the two different maskers. Specifically, masking period patterns (MPPs) for the +SCHR and -SCHR maskers were measured to show the influence of the phase relationships, and thus, the spectrotemporal characteristics of harmonic complexes on masking effectiveness. An enhanced channel model (Oh, 2013) provides a quantitative explanation for masking differences between +SCHR and -SCHR maskers by introducing nonlinear channel correlation across both frequency and time and a nonlinear decision criterion. [Research supported by a grant from the Office of Naval Research #N000140911017.]

1aPP21. Dual-carrier vocoder: Evidence of a primary role of temporal fine structure in streaming. Frederic Apoux, Carla L. Youngdahl, Sarah E. Yoho, and Eric W. Healy (Speech & Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, fred.apoux@gmail.com)

Thus far, two possible roles of temporal fine structure (TFS) have been suggested for speech recognition. A first role is to provide acoustic speech information. A second role is to assist in identifying which auditory channels are dominated by the target signal so that the output of these channels can be combined at a later stage to reconstruct the internal representation of that target. Our most recent work has been largely in contradiction with the speech-information hypothesis, as we generally observe that normal-hearing (NH) listeners do not rely on the TFS of the target speech signal to obtain speech information. However, direct evidence that NH listeners do rely on the TFS to extract the target speech signal from the background is still lacking. The present study was designed to provide such evidence. A dual-carrier vocoder was implemented to assess the role of TFS cues in streaming. To our knowledge, this is the only strategy allowing TFS cues to be provided without transmitting speech information. Results showed that NH listeners can achieve sentence recognition scores comparable to that obtain with the original TFS (i.e., unprocessed), suggesting a primary role of TFS cues in streaming. Implications for cochlear implants are discussed.

1aPP22. Modeling temporal effects in two-tone suppression. Erica L. Hegland and Elizabeth A. Strickland (Speech, Lang., and Hearing Sci., Purdue Univ., Heavilon Hall, 500 Oval Dr., West Lafayette, IN 47907, ehegland@purdue.edu)

The medial olivocochlear reflex (MOCR) reduces the gain of the active process in the cochlea. Both physiological and psychoacoustical data support the hypothesis that the MOCR may improve the response to a tone in noise. This is generally interpreted as resulting from a decrease in the response to the noise. However, elicitation of the MOCR may also reduce suppression, an instantaneous cochlear gain reduction that is a by-product of the cochlear active process. There is only limited physiological and psychoacoustic data on the effect of the MOCR on suppression. Recently, the time-course of the MOCR has been integrated into a well-established computational auditory nerve model [Smalt *et al*., J. Assoc. Res. Otolaryngol. (2013)]. The purpose of this study is to use the auditory nerve model to systematically examine suppression and the effects of the MOCR at the level of the basilar membrane and the auditory nerve. These results will provide more understanding on the interaction of these two types of gain reduction and how they relate to speech understanding in noise. [Research supported by NIH(NIDCD) R01 DC008327 and T32 DC00030.]

1aPP23. Modeling speech intelligibility in distortion conditions using a speech-based speech transmission index. Gongqiang Yu, Anthony J. Brammer, and Eric R. Bernstein (Ergonomic Technol. Ctr., Univ. of Connecticut Health Ctr., 263 Farmington Ave., MC 2017, Farmington, CT 06030-2017, gyu@uchc.edu)

A modified objective model expanding on the speech-based speech transmission index is proposed to predict speech intelligibility under various conditions of nonlinear distortion. The proposed model computes values over a time window by analyzing the signal to noise ratio and the modulation transfer function between the input and output of a transmission channel. The channel is divided into frequency bands with ranges compatible with critical bands. The cross covariance among adjacent frequency bands is also considered. The index is obtained by averaging the calculated values of these time windows. The proposed model is evaluated with subjective measurement of word intelligibility scores using the modified rhythm test with non-linear distortions of phase jitter and clipping introduced into the speech material presented to subjects. The results demonstrate that high correlations between the indices of the proposed model and the intelligibility scores are maintained for these distortions.

1aPP24. Defining essential characteristics of reliable spectral properties that elicit spectral contrast effects in vowel identification. Paul W. Anderson and Christian Stilp (Univ. of Louisville, 2329 Mount Claire Ave., Apt. 4, Louisville, KY 40217, paul.anderson@louisville.edu)

Auditory perception excels at extracting reliable spectral properties from the listening environment. Preceding acoustic contexts filtered to emphasize a narrow frequency region (Kiefte and Kluender, 2008 *JASA*), broad differences between two vowel spectra (Watkins, 1991 *JASA*), or resynthesized to shift wide ranges of frequencies (Ladefoged and Broadbent, 1957 *JASA*) all influence identification of a subsequent vowel sound. Spectral differences between filtered contexts and subsequent vowel targets were perceptually enhanced, resulting in contrast effects (*e.g.*, emphasizing spectral properties of [I] in the context produced more [ε] responses). Historically, this phenomenon has been studied using filters whose gain was broadband and/or high-amplitude, providing very strong evidence for these

reliable spectral properties. Essential characteristics of these filters that are necessary and/or sufficient to elicit spectral contrast effects are unknown. A series of experiments examined relative contributions of filter frequency, amplitude, and bandwidth to reliable spectral properties that elicit contrast effects in vowel identification. Preceding sentence contexts were processed by narrowband, broadband, or spectral envelope difference filters derived from endpoints of a vowel series differing in one frequency region (*e.g.*, F_1 in [I] and [ɛ]). Preliminary results suggest complex dependencies on filter amplitude and bandwidth for vowel identification; further results will be discussed.

1aPP25. Establishing a clinical measure of spectral-ripple discrimination. Michelle R. Molis (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, michelle.molis@va.gov), Rachael Gilbert (Dept. of Linguist, Univ. of Texas at Austin, Austin, TX), and Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., Portland, OR)

Spectral-ripple discrimination thresholds have been used effectively to assess frequency-resolving power in cochlear implant users. To improve potential clinical utility as a reliable and time-efficient measure of auditory bandwidth for listeners with acoustic hearing, possible confounds and limitations of the method must be addressed. This study examined frequency specificity and the possibility of edge-listening with narrowband stimuli. An adaptive 4-IFC procedure was used to determine ripple discrimination thresholds for normally hearing (NH) and hearing-impaired (HI) listeners. Stimuli were broadband (100–5000 Hz), high pass (1000–5000 Hz), or low pass (100–1000 Hz) logarithmically scaled, sinusoidally modulated Gaussian noises. In some conditions, Gaussian flanking noise was introduced to eliminate potential edge-listening cues. As expected, discrimination thresholds were significantly reduced for the HI listeners. Additionally, results indicate that both NH and HI listeners are able to use edge cues to improve discrimination thresholds. The introduction of flanking noise significantly reduced thresholds, and this effect was largest for the high pass stimuli. These results can be used to evaluate the usefulness of this method as a rapid and efficient means of assessing the effective frequency bandwidth of listeners with acoustical hearing. [Work supported by NIH.]

1aPP26. Sleeping position can reduce the effect of snoring on sleeping partners. Jen-Fang Yu, Yen-sheng Chen (Graduate Inst. of Medical Mechatronics, Chang Gung Univ., 259 Wen-Hwa 1st Rd., Kwei-Shan Tao-Yuan,-Taoyuan, Taiwan333, Taiwan, yhn888888@gmail.com), Hsueh-Yu Li (Otolaryngol., Chang Gung Memorial Hospital, Tao-yuan, Taiwan), and Li-Ang Li (Otolaryngol., Chang Gung Memorial Hospital, Taoyuan, Taiwan)

During sleep, the sleep partner is usually the first and direct victim of snoring. When snorers make mild snoring sounds, the majority of sleep partners will tolerate the sounds. In this study, the sound stimulus of snoring was the average snoring sound of ten snorers obtained using Adobe Audition. The sleeping position of sleep partners is either to the left or right of the snorers. Therefore, in this study, the measurements were performed in a semi-anechoic room, and positions of 0 degree to 180 degrees were used to represent the sound sources of snoring. This study performed measurements of the sound field of sleep partner's sleeping positions at three distances and at angles between 0 degree and 180 degree. The results demonstrate that the difference in volume received by the two ears was the largest at a separation distance of 30 cm and an angle of 90 degree. As mentioned in previous studies, this result occurred due to the head shadow effect being more pronounced at high frequency, while the snoring in this study was a complex sound and the frequencies were mostly low.

1aPP27. Frequency shifts in distortion product otoacoustic emission evoked by fast sweeping primaries in adults and newborns. Hammam A. AlMakadma, Beth A. Prieve (Commun. Sci. and Disord., Syracuse Univ., 261 Skytop Rd., Ste. 1200, Syracuse, NY 13210, haalmaka@syr.edu), Walid M. Dyab (L.C. Smith College of Eng. and Comput. Sci., Syracuse Univ., Syracuse, NY), Glenis R. Long, and Simon Henin (Speech-Language-Hearing Sci., Graduate School and Univ. Ctr., New York, NY)

Distortion product otoacoustic emissions (DPOAEs) are a vector sum of two components, generator and reflection, which produce overall DPOAE levels with a pattern of minima and maxima across frequency referred to as fine structure. The pattern of maxima and minima shifts higher or lower in frequency dependent on sweep direction (Henin *et al.*, 2011), consistent with cochlear scaling invariance. A break from scaling invariance occurs between 1 and 1.4 kHz in human adults. DPOAE phase at frequencies below the "break" are steeper in newborns than adults. We probed frequency shifts to up- and down-swept primaries of 1 octave/s in adults and newborns. Frequency shifts were examined for DPOAEs evoked by up-sweeps and down-sweeps using a covariate correlation function across the entire DPOAE frequency range, and above and below 2 kHz. Newborns had significantly greater frequency shifts in the reflection component below 2 kHz than adults. There was a significantly different fine structure frequency shift between the low- and high-frequency ranges in adults. In newborns, this difference in the frequency shifts for the low- and high- frequencies was significant for the reflection component. These preliminary findings suggest that frequency shifts can be used to assess potential maturational differences in cochlear mechanics.

1aPP28. The sharp frequency selectivity of low- and medium- spontaneous rate auditory nerve fibers might allow for rate-place coding up to 5 kilohertz. Marcos A. Cantu (Ctr. for Computational Neurosci. and Neural Technol. (CompNet), Boston Univ., 677 Beacon St., Boston, MA 02215, cantu@bu.edu)

One feature that might differentiate the three types of spontaneous rate (SR) auditory nerve fibers (ANFs) is the sharpness of frequency tuning at comparable sound pressure levels. My hypothesis, prior to modeling the tuning curves for each fiber type, was that low-SR fibers have a higher threshold for stimulation and thus have sharper frequency selectivity than medium-SR fibers, which in turn have sharper frequency selectivity than high-SR fibers. The results of the simulation support this framework. I used the Zilany *et al.* (2014) model of the auditory periphery and the cochlear tuning parameters from Shera *et al.* (2002) to generate tuning curves and neurogram raster plots for each of the three fiber types. The results of the simulation suggest that the sharp frequency selectivity of low-SR and medium-SR ANFs might allow for resolved rate-place coding up to 5 kHz. At frequencies below 1500 Hz, high-SR fibers were seen to have very different response properties than low-SR and medium-SR fibers. It is conceivable that the different fiber types constitute parallel pathways and mediate two different coding schemes. We should consider whether these sharply tuned low-SR and medium-SR ANFs, while fewer in number than High-SR ANFs, might be especially important for rate-place frequency coding.

1aPP29. Neural discrimination of degraded speech. Mark Steadman and Christian J. Sumner (MRC Inst. of Hearing Res., Sci. Rd., University Park, Nottingham NG14 5GL, United Kingdom, mark.steadman@ihr.mrc.ac.uk)

Cochlear implants provide a degraded input to the auditory system. Despite this, cochlear implant users are able to discriminate speech sounds in quiet with a degree of accuracy comparable to that of normal hearing listeners. The neural bases of this phenomenon is not well understood. A set of vowel-consonant-vowel phoneme sequences, each produced by multiple talkers, were parametrically degraded using a noise vocoder. Neural responses were recorded in the guinea pig midbrain and cortex, and auditory nerve responses were generated using a computational model. The discriminability of these responses was quantified using a nearest neighbor classifier. When envelope modulations were limited to 16 Hz, classifier performance was qualitatively similar to that of human listeners for all brain regions. However, in the auditory nerve and the midbrain, the preservation of high rate envelope cues enabled the near perfect discrimination of speech tokens even for heavily spectrally degraded speech. High rate envelope cues do not appear to increase discriminability of auditory cortex responses. High rate envelope cues, represented up to the midbrain, are useful for discriminating speech tokens. However, qualitatively more consistent with perception, high rate envelope cues do not contribute to the discriminability of cortical neural responses.

1aPP30. Pitch shift of the residue and its brainstem electrophysiological correlates are explained by nonlinear oscillation. Karl D. Lerud, Ji Chul Kim, and Edward W. Large (Psych., Univ. of Connecticut, 77 Forest Rd., Apt. B, Storrs, CT 06268, karl.lerud@uconn.edu)

Data show that amplitude modulation is an important factor in the neural representation and perceived pitch of sound. However, sounds with identical Hilbert envelopes can elicit different pitches. Sounds with consecutive harmonics elicit a pitch at the difference frequency of the harmonics. If this complex is shifted up or down, the amplitude envelope stays the same, but the perceived pitch moves in the direction of the shift; thus the fine structure contributes to its perceived pitch. Physiologically, auditory nerve spike timing data have shown that fine structure is preserved during mechanical to neural transduction, and brainstem EEG experiments in humans show a unique and consistent frequency-following response (FFR) associated with pitch-shifted stimuli. Here, we model the FFR with canonical networks of nonlinear oscillators representing the cochlea, cochlear nucleus, and lateral lemniscus. Dynamical analysis reveals a resonance at the frequency of the perceived pitch that is rooted in the fine structure-locked portion of the response. We show that these responses are natural outcomes of an intrinsically nonlinear system. Our results provide good evidence for highly nonlinear auditory brainstem processing, and suggest that these nonlinearities are essential for the perception of pitch.

1aPP31. Degraded temporal processing after traumatic brain injury. Eric Hoover, Pamela E. Souza (Northwestern Univ., 2240 Campus Dr., Evanston, IL 60201, EricHoover2014@u.northwestern.edu), and Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland, OR)

Hearing complaints are common following traumatic brain injury (TBI) even in the absence of peripheral auditory impairment. Study goals were to explore the mechanisms which underlie this complaint. Adult listeners with a history of mild TBI or concussion were compared to young and age-matched controls. All listeners had normal or near-normal audiometric thresholds. Listeners with TBI reported difficulty understanding speech in noise. We hypothesized that hearing complaints after TBI are related to damage to neural pathways resulting in degraded temporal resolution. A battery of psychophysical tasks were used to test this hypothesis, specifically: monaural temporal fine structure, to evaluate temporal coding of unresolved harmonics; binaural temporal fine structure, to evaluate temporal coding of a low-frequency sinusoid; interaural coherence, to evaluate sensitivity to the similarity of a correlated broadband noise; and masking level difference, to evaluate the ability to take advantage of interaural phase for tone detection in noise. Results showed individual differences in temporal processing ability among listeners with a history of TBI. Test scores varied from normal to significantly impaired, despite normal audiometry. These results suggest that auditory temporal processing may be related to difficulty understanding speech in noise after TBI. [Work supported by NIH.]

1aPP32. Optically generated responses in the cochlear nerve. Suguru Matsui (Sensory and Cognit. Neural System Lab., Graduate School of Life and Medical Sci., Doshisha Univ., 12-3 Oyanagi, Miyaketyo, shikigun, Nara 636-0216, Japan, dmn1018@mail4.doshisha.ac.jp), Kota I. Kobayasi (Sensory and Cognit. Neural System Lab., Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe-shi, Japan), and Hiroshi Riquimaroux (Neurosensing and Bionavigation Res. Ctr., Doshisha Univ., Kyotanabe-shi, Japan)

Optical stimulation has been shown to substitute the electrical stimulation because the optical method can be more spatially selective, less invasive, and generate no electrical artifact, comparing to electrical stimulation. Previous study researched that pulsed near infrared laser irradiation to cochlear nerves in Mongolian Gerbil evoked the compound action potentials (CAP) in the cochlear nerves. Purpose of this study is to examine the differences between optically and acoustically induced CAP. Wavelength of the laser was 1.871 μm. The neural response was evaluated by using following parameters, pulse width (10–1000 μs), radiant exposure (0–2.017 mW), and repetition rate (1–1000 Hz). Click sounds used for comparison had similar parameters with the laser, click sound duration (10–1000 μs), sound pressure level (35–80 peak equivalent dB SPL), and repetition rate (1–1000 Hz). A silver wire electrode was set on the bony rim of the round window. Optically induced neural activity was confirmed at 1–1000 Hz repetition rate and was synchronized to stimulation cycles of 1–1000 Hz. Increasing radiant exposure induced larger CAP amplitude. Laser stimulation could reproduce CAP amplitude in the range of acoustically induced one. It is concluded that near infrared laser irradiation generates similar neural activities as click sound does.

1aPP33. Effect of basilar and tectorial membrane properties and gradients on cochlear response. John Cormack, Yanju Liu, Jong-Hoon Nam, and Sheryl Gracewski (Mech. Eng., Univ. of Rochester, 217 Hopeman, Rochester, NY 14627, sheryl.gracewski@rochester.edu)

The cochlea is a spiral-shaped, fluid-filled organ in the inner ear that converts sound with high resolution over a large frequency range to neurological signals that can then be interpreted by the brain. The organ of Corti, supported below by the basilar membrane and attached above to the tectorial membrane, plays a major role in the amplification of small signals. In early fluid-structure interaction models of the cochlea, the mechanical properties of the organ of Corti were neglected and only the basilar membrane was considered, approximated by a series of springs. Recent experiments suggest that the mechanical properties and property gradients of the tectorial membrane may also be important for frequency response of the organ of Corti and that separate waves may propagate along the basilar and tectorial membranes. Therefore, a two-dimensional two-chamber finite difference model of the cochlea was developed to investigate the independent responses of the basilar and tectorial membranes. Responses are compared for models using one-, two-, or three-degree-of-freedom approximations for the organ of Corti, with parameters derived from a physiologically based finite element model. The effects of independent coupling of the fluid to the tectorial and basilar membranes and longitudinal coupling along the membranes are investigated.

Session 1aUW

Underwater Acoustics: Acoustic Tomography and Interferometry

John R. Buck, Chair
ECE, UMass Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747

Contributed Papers

10:00

1aUW1. Acoustic noise interferometry in the Straits of Florida at 100 m depth: A ray-based interpretation. Oleg A. Godin, Nikolay A. Zabotin, Liudmila Zabotina, Justin S. Ball (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO 80305-3328, oleg.godin@noaa.gov), Michael G. Brown, and Neil J. Williams (RSMAS, Univ. of Miami, Miami, FL)

Cross-correlation function of ambient and shipping noise recorded simultaneously by two hydrophones provides an estimate of the acoustic Green's function, which describes deterministic sound propagation between the hydrophones and can be used to estimate physical parameters of the water column and seafloor. This paper presents results of an experimental investigation of acoustic noise interferometry in 100 m-deep water in the Straits of Florida. Acoustic noise was recorded continuously for about six days at three points near the seafloor. Coherent acoustic arrivals are successfully identified in the 20–70 Hz frequency band for pairs of hydrophones separated by ranges of 5.0 and 9.8 km. The measured noise cross-correlation functions are compared to ray-based simulations of Green's functions, with generally good agreement between correlation functions and simulations. Ray-based simulations are shown to reproduce multipath features of the measured correlation functions, which are due to multiple surface and bottom reflections. The feasibility of passive acoustic remote sensing using a few hydrophones in a shallow-water waveguide will be discussed. [Work supported by NSF, ONR, and NAVAIR.]

10:15

1aUW2. Acoustic noise interferometry in the Straits of Florida at 100 m depth: A mode-based interpretation. Michael G. Brown, Neil J. Williams, Geoffrey Banker (RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, mbrown@rsmas.miami.edu), Oleg A. Godin, Nikolay A. Zabotin, and Ludmilla Zabotina (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO)

A field experiment designed to test the feasibility of noise interferometry was conducted in the Florida Straits in December 2012 in water of approximately 100 m depth. Ambient noise was recorded concurrently on three moored near-bottom instruments with horizontal separations of approximately 5 km, 10 km, and 15 km. Consistent with theoretical predictions, coherent sums (stacks) of many realizations of ambient noise at two measurement locations are shown to yield approximations to deterministic Green's functions that describe propagation between the two locations. Band-pass filtering of measured stacked cross-correlations reveal modal dispersion characteristics of the estimated Green's functions that are compared to mode-based simulations. [Work supported by NSF and ONR.]

10:30

1aUW3. Acoustic noise interferometry in the Straits of Florida at 600 m depth: Preliminary results. Michael G. Brown, Neil J. Williams, Xiaoqin Zang (RSMAS, Univ. of Miami, 4600 Rickenbacker Cswy, Miami, FL 33149, mbrown@rsmas.miami.edu), Oleg A. Godin, Nikolay A. Zabotin, and Ludmilla Zabotina (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO)

A field experiment designed to test the feasibility of noise interferometry was conducted in the Florida Straits in September/October 2013 in water of approximately 600 m depth. Ambient noise was recorded concurrently on three moored near-bottom instruments with horizontal separations of approximately 5 km, 10 km, and 15 km. Consistent with theoretical predictions, coherent sums (stacks) of many realizations of ambient noise at two measurement locations are shown to yield approximations to deterministic Green's functions that describe propagation between the two locations. Ray- and mode-based simulations of approximations to Green's functions are compared to measured stacked cross-correlations. [Work supported by NSF and ONR.]

10:45

1aUW4. Shear-wave velocity estimation from different type of sediments. Hefeng Dong (Dept. of Electronics and Telecommunications, Norwegian Univ. of Sci. and Technol., NTNU, Trondheim NO-7491, Norway, hefeng@iet.ntnu.no)

This paper presents estimates of shear-wave velocity profiles for different types of sediments by inverting the dispersion curves of interface waves. Data used in this study were collected at different locations with different recording devices in different experiments. Four data sets were analyzed. Data from one set were recorded by hydrophone array, and data from other three sets were collected by multi-component ocean bottom seismic cables. The configuration of the experimental setup and the environmental conditions were different for the different experiments. Water depth varied from 2 m to 365 m. The frequency range of the interface-wave data was from 1.5 Hz to around 20 Hz. The phase-velocity dispersion curves of the interface waves were extracted using different time-frequency analysis methods. The maximum penetration depth of the interface waves to the sediments varied from 15m to 180 m. A Bayesian nonlinear inversion approach was used for estimating shear-wave velocity profiles as a function of depth in the sediments and the uncertainties. The estimated shear-wave velocity profiles from different experiment and different sediment type were discussed and compared.

11:00

1aUW5. Wavenumber analysis of interface wave characteristics using elastic parabolic equation solutions. Scott D. Frank (Marist College Mathematics, Marist College, 3399 North Ave., Poughkeepsie, NY 12601, scott.frank@marist.edu), Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, Golden, CO), and Robert I. Odom (Appl. Phys. Lab., Univ. of Washington, Seattle, WA)

Interface waves that travel along the ocean bottom are known as Scholte waves and can contribute to the deep ocean acoustic field, even at depths below the ray-theoretical turning point. Scholte waves spread by the inverse of the square root of range and propagate greater distances along the ocean floor than ocean acoustic energy. Elastic parabolic equation solutions are effective for analysis of Scholte wave behavior with respect to environmental parameters since these waves represent interactions between dilatational and rotational elastic waves resulting from elastic boundary conditions. Generation of interface waves by water column and buried seismic sources will be demonstrated. Hankel transforms of calculated acoustic pressure will be used to evaluate impact of elastic parameters on interface wave amplitudes and ducting effects in elastic sediment layers. The effect of large-scale bathymetry on interface wave propagation will also be investigated. [Work supported by ONR.]

11:15

1aUW6. Implementing physical constraints for noise-only mode shape estimation on real data. Ian M. Rooney, John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts at Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, irooney@umassd.edu), and Kathleen E. Wage (Elec. and Comput. Eng., George Mason Univ., Fairfax, VA)

Many underwater acoustic tasks sample a narrowband pressure field with a vertical line array. In the absence of strong local sources, the noise sampled by an array includes both a spatially correlated and a spatially uncorrelated component. The acoustic waveguide's modes form a basis for the spatially correlated noise component generated by distant sources. This basis can be estimated from the eigenvectors of the noise sample covariance matrix. Propagation physics constrain the mode shapes to be real, but the eigenvectors are generally complex. These complex vectors require a phase rotation for each eigenvector prior to taking the real part. Previous work found that the eigenvector mode estimates' phases include spatially correlated noise. This noise correlation degrades the performance of spatial averaging and minimum variance estimates of the unknown phase rotation. The best estimate is the phase of the array element with the maximum magnitude for each mode, which is robust to the noise's spatial correlation. This research evaluates the previously developed phase rotation techniques using a real data set. The data was collected off the north coast of Elba Island by the SACLANT Centre in 1993. [This work was supported by ONR.]

11:30

1aUW7. Remote nautical bottom estimation for the safety of navigation offshore Amazon River mouth. Qunyan Ren and Jean-Pierre Hermand (Acoust. & Environ. HydroAcoust. Lab., Labs. of Image, Signal Processing and Acoust.-Ecole polytechnique de Bruxelles / Faculté des Sci., U.L.B., ave. F.-D. Roosevelt 50, CP 165/57, Brussels 1050, Belgium, qunyren@ulb.ac.be)

In muddy areas, the fluid sediment layer can reach a level of "nautical bottom" that can contact with a ships keel causing either damage or unacceptable effects on controllability and maneuverability. Consequently, the minimum depth and allowed draught need to be determined for the safety of navigation. An acoustic remote sensing technique is proposed to facilitate navigation safety applications through the determination of sediment layer properties. It uses range and frequency-dependent features of the vertical waveguide characteristic impedance, as defined by the ratio of pressure and vertical particle velocity (or pressure gradient) at a given frequency. Such ratio can circumvent inversion uncertainty due to insufficient knowledge of complex time-varying ship noise spectrum because of its source spectral level independent. Real ship noise data recorded on a compact array offshore at the mouth of the Amazon River in Brazil, 2012, are processed by a global optimization based inversion scheme. The inverted results demonstrate that technique can estimate the effective water depth through resolving sediment layers properties, especially the density. The promising results demonstrate the feasibility of this technique to facilitate safety navigation applications at port areas.

11:45

1aUW8. Sound speed and attenuation measurements of sandy sediments in reverberation chamber. Qi Li, Qi S. Yu, and Wang Y. Huang (Acoust. Sci. and Technol. Lab., Harbin EngineeringUniv., No.45 Nantong St., Nangang District, Harbin, Heilongjiang, Harbin, Heilongjiang, China, liqi@hrbeu.edu.cn)

Sound speed and attenuation in sandy sediments are important acoustic parameters. But the uncertainties of current *in-situ* measurements at low-frequency are very large and the data are not sufficient to be used to test theory predictions. As an alternative, measurements in a water filled isolated reverberation chamber were attempted in laboratory, which preserves lower frequency limitation and smaller scale requirement of water tank. In order to know the feasibility of the method, a type of fine sand sediments was measured. Spatial averaged reflection and transmission coefficients of the sand layer and attenuation were calculated over the frequency range of 90–170 kHz. And sound speed was inverted with these measured parameters. After then measurements at low-frequency till to 2 kHz were tried in the same tank.

Session 1pAA

Architectural Acoustics: Room Acoustics Prediction and Measurement

Timothy Foulkes, Chair
Cavanaugh Tocci Assoc., 327 Boston Post Rd., Sudbury, MA 01776

Chair's Introduction—1:00

Contributed Papers

1:05

1pAA1. Audio/visual concepts for human/robot communication in immersive virtual environments. Jonas Braasch, Richard J. Radke (Ctr. for Cognition, Commun., and Culture, Rensselaer Polytechnic Inst., 110 8th, Troy, NY 12180, braasj@rpi.edu), John Wen (Ctr. for Automation Technologies and Systems, Rensselaer Polytechnic Inst., Troy, NY), Mei Si, Andrew Cunningham (Ctr. for Cognition, Commun., and Culture, Rensselaer Polytechnic Inst., Troy, NY), William Keddy-Hector, and Utkarsh Sinha (Ctr. for Automation Technologies and Systems, Rensselaer Polytechnic Inst., Troy, NY)

Communication between robots and humans is a challenge in complex built environments. In this project, we are exploring how previous achievements in this area apply to human/robot communication within immersive virtual displays. In our concrete example, we host human-scale robots that interact with humans in our Collaborative-Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab). The CRAIVE-Lab provides a physical/digital environment for collaborative tasks using seamless video from multiple projectors and a 128-channel wave-field system. The system is designed to monitor the whole floor-space area (10 m $\times$ 12 m) with a camera network mounted to the ceiling grid and a microphone array combining shotgun, spherical and possibly people-worn microphones. Based on the sensor data, a visual analysis and an auditory scene analysis are performed. The latter includes sound localization, speech and musical-feature recognition. The robots, a Rethink Baxter robot mounted on an electric wheelchair and a Robokind Zeno, are used to perform assistive technology and social communication tasks. Both robots have direct access to these data using a digital feed over a wireless network to augment their own sensor systems of built-in cameras and microphones. [Work supported by NSF grant #1229391.]

1:20

1pAA2. Comparison of computational predictions of broadband interior sound fields with experimental measurements. Krista A. Michalis (Structural Acoust. and Target Strength, NSWC Carderock, 9500 MacArthur Blvd., Bldg. 3 Rm. 341B, West Bethesda, MD 20817, krista.michalis@navy.mil) and Donald B. Bliss (Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC)

Predictions of broadband interior noise using an energy-intensity boundary element method (EIBEM) are compared to experimental measurements. Steady-state sound fields with reflection boundaries are modeled with the EIBEM method. The specular reflection field is represented using uncorrelated broadband directional sources, expressed as spherical harmonics. For each boundary element, the amplitudes of the harmonics are determined from the incident field from all other elements and sources, and are subject to energy conservation using a Lagrange multiplier integral constraint. The computational solution utilizes a relaxation method starting with 3-D diffuse reflection. Earlier EIBEM results were compared to exact analytical solutions obtained from modal analysis, and to a broadband image method. For the experimental study, 1/3-octave band measurements were made within a full-scale steel-walled enclosure with different absorption levels, distributions, and source locations. Measurements were made along three 10-microphone linear arrays spanning the enclosure, and by microphones at other selected locations. Source power was measured independently, and random incidence absorption was inferred from both reverberation times and from steady state processing using multiple microphone locations. Predicted and measured spatial variation of the steady-state fields were in good agreement. The sensitivities of the predictions to various assumptions, geometric, anomalies, and experimental uncertainties are discussed.

1:35

1pAA3. Modeling the effects of air currents on acoustic measurements in large spaces. David H. Griesinger (Res., David Griesinger Acoust., 221 Mt Auburn St. #504, Cambridge, MA 02138, dgriesinger@verizon.net)

Historically reverberation times were accurately measured with an explosive or an interrupted tone or noise. But in theory using convolution methods for acoustic measurement requires that an impulse response be precisely time stationary. Air in large spaces is never stationary, and the further a sound wave travels—as it must in a reverberant tail—the more it will be affected by moving air. Long stimuli and synchronous averaging increase the time over which the air must be still. We propose that air currents can be modeled through the fluctuations they create in the time base of the received stimulus. We find that noise-based convolution methods such as MLS are particularly sensitive to this problem, as they must use long stimuli to achieve an adequate signal-to-noise ratio (S/N). For these methods, when there is no net flow, the principle degradation is a rapid reduction of S/N at high frequencies. If there is a net flow, the high frequency reverberation time can also be significantly shortened. Logarithmic sine stimuli are shorter than MLS for an equivalent S/N, and are less sensitive to air currents. The relevance of these effects in actual spaces is under investigation.

1:50

1pAA4. The position of the sound source in churches. Umberto Berardi (Civil and Environ. Eng. Dept., Worcester Polytechnic Inst., via Orabona 4, Bari 70125, Italy, u.berardi@poliba.it) and Francesco Martellotta (DICAR, Politecnico di Bari, Bari, Italy)

The position of the sound source is an aspect rarely considered in room acoustics because the shape of the space and its internal organization generally make clear where the sound source should be. However, this may not be the case in some buildings, such as churches. In these buildings, the source is not located in one position, but it moves from choir to the altar or the pulpit. This makes problematic the analysis of church acoustics. In particular, if the reverberation time is almost constant whenever the source may be, other acoustic parameters strongly depend on the position of the sound source. This paper focuses on the effect of the position of the sound source over the early lateral fraction and the center time. First, the results of computer simulations allow discussing the influence of some locations of the organ: above the entrance, in the middle of the room, between the altar and the worshipers, and behind the altar. Then, the values of the acoustic parameters for different sound sources are evaluated. In particular, the effect of speaking from the pulpit or from the altar are compared. Finally,

suggestions to evaluate several positions of different sound sources in churches are reported.

2:05

1pAA5. Research on the parametric design of concert hall and its acoustics. Lu W. Shuai and Xiang Yan (Acoust. Lab., School of Architecture, Tsinghua Univ., Rm. 104, Main Bldg., Beijing 100084, China, 359901298@qq.com)

Concert hall is a complex system containing different building elements and technical issues such as acoustics, sight, circulation, ventilation, etc. It takes designers a lot of efforts to modify the design or test its technical performance when using traditional design method. We explored the possibilities of applying parametric design method to concert hall design, developing an interactive parametric tool which could modify all the building elements according to designers' desires and functional needs, reducing designers' work burden; meanwhile, it could simultaneously show the acoustic attributes of the design, aiding to acquire a satisfactory technical performance. This paper introduces the basic framework of the parametric tool and also shows an example to demonstrate its validity.

2:20–2:35 Break

2:35

1pAA6. Measuring sonic presence and the ability to sharply localize sound using existing lateral fraction impulse response data. David H. Griesinger (Res., David Griesinger Acoust., 221 Mt Auburn St. #504, Cambridge, MA 02138, dgriesinger@verizon.net)

The ability to sharply localize sound sources creates a perception of presence that plays a large role in grabbing and holding attention, which is of major importance in learning and performance venues. The ability to localize accurately depends critically on how the audibility of the direct sound is reduced by reflected energy arriving in the first ~80 ms. Data analysis methods that quantify the ability to localize sound at a given seat from impulse response data are being developed—but binaural impulse response data is not common in acoustic data bases. Human hearing is not omnidirectional. It uses both ILD and ITD differences between the two ears to localize frontal sound sources with an acuity of about two degrees—far more accurately than current first-order microphones. Head shadowing also significantly lowers the strength of lateral reflections that arrive at each ear. This paper presents a method whereby existing data from omnidirectional and figure-of-eight microphones can be manipulated to emulate the ILD, ITD, and head shadowing of a binaural microphone. The method uses spherical harmonics and HRTF data to restore head shadowing and estimate the ITD and ILD of the direct sound and the first few reflections.

2:50

1pAA7. Research on the measurement and application of surface diffusivity—Takes Concert Hall at Gulangyu Music School in Xiamen as an example. Lu W. Shuai, Xiaoyan Xue, Peng Wang, and Xiang Yan (Acoust. Lab, School of Architecture, Tsinghua Univ., Rm. 104, Main Bldg., Beijing 100084, China, 359901298@qq.com)

The surface diffusivity is an important acoustical attribute of material, which is highly correlated with the sound quality of a concert hall. This paper introduces the theory and practical method to measure the surface diffusivity in laboratory, which is more precise and objective than traditional Surface Diffusivity Index (SDI), and then takes the design of the concert hall at Gulangyu Music School in Xiamen as an example, shows the diffusivity features of its six-sided-cylinder shaped GRG ceiling material, demonstrates the design method and significance of using diffusive material in room acoustic design.

3:05

1pAA8. The use of sport centers for musical shows. Amelia Trematerra (Dept. of Architecture and Industrial Design, Second Univ. of Naples, borgo san lorenzo, Aversa 83016, Italy, amelia.trematerra@unina2.it) and Gino Iannace (Dept. of Architecture and Industrial Design, Second Univ. of Naples, Aversa, Caserta, Italy)

The sport centers are often used for musical shows, these buildings are built for sporting events, and they have a large volume with surfaces acoustically reflective (concrete steps, plaster and concrete ceilings) and have high values of reverberation time. In this work, a study of sporting center "Pala Jacazzi" in Aversa (Italy) for musical shows is reported. This building was designed for sport such as basketball and volley, has a volume of 28.000 cubic meters, length is 60 m, width is 60 m, and the seating capacity is 2.000. The acoustic measurements were carried out in accordance with ISO 3382, with an omnidirectional sound source, feed with a MLS signal, put in the game rectangle, and the measurements microphone put in different point on the steps. The average value, without audience, of T30 at 1.0 kHz is over 5.0 s; this is an high value if the sporting center must be used for musical shows. The software for the architectural acoustic "Odeon" is used to choose the type of absorbent material to insert and to evaluate the area of the walls and ceiling to cover for reduce T30 values, without audience, to 3.0 s.

3:20

1pAA9. Mixing console design for telematic applications in live performance and remote recording. David J. Samson and Jonas Braasch (Rensselaer Polytechnic Inst., 1521 6th Ave., Apt. 303, Troy, NY 12180, samsod2@rpi.edu)

The purpose of this presentation is to introduce a hardware-based mixing console architecture for telematic applications that integrates key features germane to distributed performance and remote recording. The current practice in state of the art telematic performance uses simple software-based interconnection with complex routing schemes and offers minimal flexibility or control over key parameters necessary to achieve a professional workflow. In addition to all customary features, the console will have surround panning capability for both the motorized binaural manikin as well as all sources within the auralization module. Key features such as self-labeling channel strips, onboard latency monitoring, synchronized remote audio recording and monitoring, and a highly flexible routing architecture will be integrated into the console design. This console design will provide a platform for the audio engineer to realize the full potential of telematics for networked performance and remote recording.

Session 1pABa

Animal Bioacoustics and Psychological and Physiological Acoustics: Comparative Perspectives on the Cocktail Party Problem II

Mark Bee, Cochair
Dept. of Ecology and Evolutionary Biology, Univ. of Minnesota, 100 Ecology, 1987 Upper Buford Circle, St. Paul, MN 55108

Micheal L. Dent, Cochair
Psychology, Univ. at Buffalo, SUNY, B76 Park Hall, Buffalo, NY 14260

Invited Papers

1:00

1pABa1. Individual differences revealed by the challenges of listening in a complex, crowded scene. Hari M. Bharadwaj (Dept. of Biomedical Eng., Boston Univ., 19 Euston St., 1B, Brookline, MA 02446, hari@nmr.mgh.harvard.edu) and Barbara G. Shinn-Cunningham (Ctr. of Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA)

To extract content and meaning from a single source of sound in a quiet background, the auditory system can use a small subset of a very redundant range of spectral and temporal features. In stark contrast, communication in a complex, crowded scene places enormous demands on the auditory system. Spectrotemporal overlap between sounds reduces modulations in the signals at the ears and causes masking, with problems exacerbated by reverberation. Often, the more sensitive neurons in the early auditory pathway are driven to saturation, yet precise sensory representation is crucial for the ability to extract spatial, pitch, and cues that support source segregation and selective attention. Consistent with this idea, many patients seeking audiological treatment seek help precisely because they notice difficulties in environments requiring auditory selective attention. Consistent with this, in the laboratory, even listeners with normal hearing thresholds exhibit vast differences in the ability to selectively attend to a target. Here, we highlight the issues faced by the auditory system in a complex scene and describe recent behavioral and electrophysiological findings that hint at the mechanisms underlying individual differences in the ability to communicate in such adverse situations.

1:20

1pABa2. Change detection in complex acoustic scenes. Maria Chait (Ear Inst., Univ. College London (UCL), 332 Gray, London WC1X 8EE, United Kingdom, m.chait@ucl.ac.uk)

The notion that sensitivity to temporal regularity (TR) plays a pivotal role in auditory scene analysis (ASA) has recently garnered considerable attention. Nevertheless, evidence supporting a primary role for TR is based on experiments employing simple stimuli consisting of one, or two, concurrent sound sequences. Whether the role of TR in mediating ASA is robust to more complex listening environments is unknown. The present study investigates sensitivity to TR in the context of a change detection task, employing complex acoustic scenes comprised of up to 14 concurrent auditory objects. Sequences of sounds produced by each object were either temporally regular (REG) or irregular (RAND). Listeners had to detect occasional changes (appearances or disappearances of an object) within these “soundscapes.” Listeners’ performance depended on the TR of both the changing object and the scene context (TR of other objects in the scene) such that RAND contexts were associated with slower response times and substantially reduced detection performance. Therefore, even in complex scenes, sensitivity to TR is critical to our ability to analyze and detect changes in a dynamic soundscape. Importantly, the data reveal that listeners are able to acquire the temporal patterning associated with at least 14 concurrently presented objects.

1:40

1pABa3. Behavioral and neuronal sensitivity concerning objective measures of auditory stream segregation. Georg Klump (Cluster of Excellence Hearing4all, School of Medicine & Health Sci., Oldenburg Univ.,Oldenburg 26111, Germany, georg.klump@uni-oldenburg.de), Lena-Vanessa Dollezal, and Naoya Itatani (Animal Physiol. & Behaviour Group, Dept. for Neurosci., Oldenburg Univ., Oldenburg, Germany)

To evaluate possible mechanisms of auditory stream segregation, it is desirable to directly compare perceptual stream segregation and its neurophysiological correlate. Objective measures of stream segregation, i.e., measures of perceptual sensitivity that differ between conditions in which one integrated stream or two segregated streams are perceived, lend themselves to the study of such phenomena. They can be applied both in studies involving human subjects and in animal studies on auditory streaming. On the one hand, we present results from a study on informational masking in Mongolian gerbils, in which better performance in an intensity discrimination task is observed if streams of target signals and distractors are processed in separate streams. On the other hand, we present results from a temporal pattern discrimination experiment in European starlings, in which better performance is achieved if the target signals and signals useful for temporal reference are processed within one stream rather than in separate streams. In both experiments we apply signal-detection theory both to the analysis of perception evaluated in behavioral experiments and to the analysis of neuronal response patterns on the midbrain and cortical level. The neuronal population response was observed to be well correlated with the behavioral sensitivity, which can shed light on the mechanisms underlying auditory stream segregation.

2:00

1pABa4. Performance-based and subjective measures of perceptual organization in humans. Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Measures of auditory perceptual organization in humans have traditionally relied on subjective reports from subjects. For instance, in alternating tone sequences, subjects might be asked to report whether they hear one or two streams. In order to investigate perceptual organization in non-human species, it can be useful to devise tests that are more objective, in that they have a right and a wrong answer and do not rely on introspection. Recent studies from our laboratory have employed various objective and subjective tasks to provide converging evidence in the search for the underlying principles of auditory perceptual organization. It is suggested that different tasks can bias potentially multi-stable percepts in one way or another, which in turn may be useful in uncovering neural correlates of perceptual organization that can vary even when the acoustic stimuli remain the same. [Work supported by NIH grant R01DC007657.]

2:20–2:40 Break

2:40

1pABa5. Cortical processes for navigating complex acoustic environments. Shihab Shamma (Univ. of Maryland, AV WIlliams Bldg., College Park, MD 20742, sas@umd.edu)

Humans and other animals can attend to one of multiple sounds and follow it selectively over time. The neural underpinnings of this perceptual feat are the object of extensive investigations. In this talk, it is argued that source segregation depends primarily on temporal coherence between responses that encode various features of a sound source. An algorithm for implementing this process will be discussed, especially the components that are inspired by adaptive auditory cortical mechanisms. The postulated necessary role of attention in this process will be addressed, and in particular how it leads to binding of all temporally coherent features of a source (e.g., timbre and location) to segregate them from the incoherent features of other sources.

Contributed Paper

3:00

1pABa6. Manatee hearing and sound localization can help navigate noisy shallow waters and cocktail events, no Lombard's needed. Edmund R. Gerstein and Laura Gerstein (Psych., Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33486, gerstein2@aol.com)

Simultaneous masking procedures were used to measure the hearing and underwater sound localization abilities of West Indian manatees. Auditory detection thresholds of pulsed, no-pulsed pure tones, and complex sounds were measured against white noise using forced-choice paradigms. Auditory thresholds as a function of intensity, center frequency, bandwidth, pulse rate, and spectral characteristics were measured. Resulting critical ratios for pure tone measurements demonstrated manatees have acute frequency filtering abilities compared with humans and other marine mammals. Signal repetition rate along with amplitude and frequency modulation providing temporal contrasts against aperiodic background noise and lowered detection thresholds. FM signal detection thresholds were measured at or below ambient background levels. Results with species specific calls and boat noise suggested loudness summation across distant critical bands, as well as FM and amplitude modulation reduced the masking effects observed with pure tones. Manatees are well adapted for hearing and locating high frequency sounds in noisy shallow water habitats where physical boundary and near surface phenomena impede the propagation of low frequencies. Narrow critical bands and selective perception of pulsed signals may be adaptations for detecting and locating species-specific vocalizations. High frequency harmonics in manatee calls provide essential directional cues between mothers and calves.

Session 1pABb

Animal Bioacoustics and Signal Processing in Acoustics: Dynamics of Biosonar: Time-Dependent Adaptations in Transmission, Reception, and Representation in Echolocating Animals I

Laura Kloepper, Chair
Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02906

Invited Papers

3:15

1pABb1. Hearing sensation change with loud sound warning in the false killer whale. Paul E. Nachtigall (Hawaii Inst. of Marine Biology, Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, nachtiga@hawaii.edu) and Alexander Y. Supin (Severtsov Inst., Russian Acad. of Sci., Moscow, Russian Federation)

Work on hearing during echolocation has demonstrated that a whale was capable of changing its hearing sensitivity while it echolocated, perhaps to protect its hearing from its own intense emitted pulses. Would a whale similarly change hearing sensitivity when warned prior to receiving a loud sound? Hearing was measured using auditory evoked potentials (AEP). The whale had been trained to station within a hoop while wearing surface electrodes. Baseline AEP dependence on test-sound level and an auditory threshold were first established for a 20 kHz tone. In a second phase, the test sound was followed by a sudden increase in amplitude up to 170 dB re 1 μPa. Thus, the faint test sounds took on the role of a warning signal for the ensuing loud (unconditioned) sound. After a few trials, the test stimuli revealed a substantial reduction of hearing sensitivity before the loud sound. When the delay between the warning tone onset and loud tone was short (varied randomly from 1 to 9 s), the whale increased its hearing thresholds (reduced sensitivity) by around 13 dB. The data indicate that: (1) the whale learned to change hearing sensitivity when warned that the loud sound was about to arrive, and (2) the learning acted only when warnings were immediate.

3:35

1pABb2. Investigating the temporal dynamics of dolphin biosonar using phantom echoes and auditory evoked potentials. James J. Finneran (US Navy Marine Mammal Program, 53560 Hull St., San Diego, CA 92152, james.finneran@navy.mil), Jason Mulsow, and Dorian S. Houser (National Marine Mammal Foundation, San Diego, CA)

Phantom echo generation replaces physical targets with electronic signals that can be delayed in time, scaled in amplitude, and broadcast to an echolocating animal to simulate echoes from distant objects. Compared to physical targets, phantom echoes have the advantages of allowing for instantaneous changes in target characteristics, independent manipulation of echo delay and echo amplitude, and easy randomization of target range. At the Navy Marine Mammal Program in San Diego, California, phantom echo generation is combined with measurements of auditory evoked potentials to investigate the temporal dynamics of biosonar signal emission and reception in bottlenose dolphins. The studies are primarily focused on examining automatic gain control mechanisms by measuring changes in hearing sensitivity — assessed via the auditory steady-state response (ASSR) to an amplitude modulated tone — over time courses corresponding to single biosonar click-echo pairs. Results show the ASSR amplitude initially decreases at the time of click emission and then recovers following click emission, with the time course of recovery related to target range, click amplitude, and tone frequency. Additional studies are focused on dynamic changes in click emissions that occur with changes in target range. [Work funded by SSC Pacific Naval Innovative Science and Engineering (NISE) program.]

3:55

1pABb3. Mechanisms of distance-invariant recognition of target strength in the biosonar of odontocetes. Alexander Supin (Inst. of Ecology and Evolution, Russian Acad. of Sci., 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru) and Paul Nachtigall (Marine Mammal Res. Program, Hawaii Inst. of Marine Biology, Kailua, HI)

Invariant target recognition by sonar requires assessment of the target strength invariably of distance despite wide variation of the echo level. Information on the echo delay and the level of the emitted pulse allows computing of the target strength. The question is: which particular mechanisms perform this computation in the biosonar of odontocetes? Investigations of the auditory evoked potentials (AEPs) during echolocation in several odontocetes have shown that comparison of the emitted pulse level, echo delay, and echo level is based on the forward masking of the echo-response by the preceding self-heard emitted click. Prolongation of the echo delay results in releasing of the echo-related AEP from masking. This release from masking compensated for the echo attenuation with distance. As a result, the echo-related AEP amplitude depended on the target strength and little depended on the target distance. Moreover, the forward-masking duration depended on the level of the emitted pulse. As a result, the echo-related AEP featured amplitude little depended on the level of the emitted pulse. The constancy of the echo-related AEP amplitude indicates the dependence of sensation level of the echo only on the target strength and the independence of both emitted click level and target distance.

4:15

1pABb4. Biosonar auditory model for target shape perception and clutter rejection. James A. Simmons (Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912, james_simmons@brown.edu), Michaela Warnecke (Psychol. and Brain Sci., Johns-Hopkins Univ., Baltimore, MD), and Jason E. Gaudette (NUWCDIVNPT, NAVSEA, Newport, RI)

Big brown bats (*Eptesicus fuscus*) emit widely beamed biosonar sounds that contain two prominent harmonics (FM1, FM2). They exploit the ripple spectrum of multiple-glint echoes to perceive target shape and the relative weakening of higher harmonic FM2 frequencies in lowpass echoes from the surrounding scene to suppress clutter. Delay discrimination experiments using electronically generated echoes show that (1) flat-spectrum or ripple-spectrum echoes are perceived as having sharply focused delay images, (2) lowpass filtering causes defocussing of perceived delay, and (3) masking release occurs between defocused images of lowpass echoes and focused images of full-band echoes. A time-frequency auditory model for the decomposition of FM echoes into temporal and spectral dimensions, followed by reconstitution of object images in terms of delay alone accounts for both shape perception and the release from clutter masking. The model's structure resembles a neuronal spectral pattern-recognizing network grafted onto a neuronal delay-line/coincidence-detection delay-determining network. The categorical segregation of focused target shape from defocused clutter emerges from a novel anticorrelation process that depends on the auditory system's restricted range of lateral interactions across adjacent frequencies. [Work supported by ONR, NSF, BIBS, and JSPS.]

4:35

1pABb5. Biosonar dynamics in horseshoe and old-world leaf-nosed bats. Rolf Mueller, Anupam K. Gupta (Mech. Eng., Virginia Tech, 1075 Life Sci. Cir, Blacksburg, VA 24061, rolf.mueller@vt.edu), Weikai He (Shandong Univ.-Virginia Tech Int.. Lab., Shandong Univ., Jinan, Shandong, China), and Mittu Pannala (Mech. Eng., Virginia Tech, Blacksburg, VA)

Horseshoe bats (family Rhinolophidae) have evolved a capable biosonar system to allow the pursuit of prey amid dense vegetation that produces large amounts of clutter echoes. Horseshoe bats have long been known to employ dynamic effects such as Doppler effect compensation and large-scale rotations of their pinnae to realize these capabilities. Recent research has produced evidence of even more pervasive dynamical biosonar properties in horseshoe bat biosonar as well as in the related Old World leaf-nosed bats (family Hipposideridae). On the emission side, these animals employ elaborate baffle shapes that surround the exit points of their ultrasonic pulses (nostrils). During echolocation, multiple parts of these noseleaves such as the anterior leaf and the lancet in horseshoe bats are set in motion, typically in synchrony with pulse emission, hence creating a time-variant channel for the exiting wave packets. Similarly, the pinnae which diffract the incoming echoes on the reception side are frequently in motion while echoes impinge on them. These motions include non-rigid changes in shape. All these effects add a dynamic dimension to the interface of the bats' biosonar with the external world, which could allow the animals to enhance the quantity and quality of the sensory information they receive.

4:55

1pABb6. Encoding phase information is critical for high resolution spatial imaging in biosonar. Jason E. Gaudette (Code 8511, Naval Undersea Warfare Ctr., 1176 Howell St., B1371/3, Newport, RI, jason.e.gaudette@navy.mil) and James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI)

The auditory system is responsible for translating acoustic information into a robust neural representation. In echolocating mammals, precise timing of neural onset-responses is critical to reconstruct complex acoustic scenes. Phase is often ignored in transmit and receive beam patterns, but it holds significance when considering broadband signals. A beam pattern's phase alternates outside of the main lobe, which leads to a frequency-dependent structure that is useful for spatial localization. For imaging in azimuth, binaural spectral patterns and time delay between the ears encode angular position. Imaging in elevation relies principally on specific spectral patterns encoded by each ear. The additional phase information decorrelates broadband echoes arriving from off-axis. This decorrelation only occurs on the order of a single wave period; however, the pattern of dispersion across frequency is sufficient to defocus echoes arriving from off-axis in the peripheral region, while accepting echoes arriving from the focal area of attention. We propose that acoustic spectral pattern matching by echolocating animals includes both magnitude and phase components of beams in the form of timing the onset response. Computational modeling results are presented showing how encoding phase information leads to high-resolution images despite dynamic environments and variability in the target strength of objects.

5:15

1pABb7. Reconstructing the acoustic scenes encountered by free-flying, foraging bats. Wu-Jung Lee (Inst. for Systems Res., Univ. of Maryland, 1147 Biology/Psych. Bldg., College Park, MD 20742, wjlee@umd.edu), Sonja Sändig, Annette Denzinger, Hans-Ulrich Schnitzler (Animal Physiol., Inst. of Neurobiology, Univ. of Tbingen, Tbingen, Germany), Timothy K. Horiuchi, and Cynthia F. Moss (Inst. for Systems Res., Univ. of Maryland, College Park, MD)

Aerial insectivorous bats face the challenge of efficient echo scene analysis for localizing obstacles and capturing prey in flight. Data collected with a telemetry microphone mounted on the foraging bat's head provide a valuable opportunity to reconstruct acoustic scenes comprised of echoes returning to the bat's ears. This study explores the information embedded in echoes from a tethered insect and background clutter recorded by the telemetry microphone in laboratory experiments. Using images from high-speed video cameras and recordings from a far-field microphone array, angular information about different objects in the echoes are restored by assimilating the reconstructed bat's position and echolocation beam aim with respect to the objects along its flight trajectory toward prey capture. This procedure is further augmented by theoretical simulations using acoustic scattering principles to circumvent the limitation imposed by the sensitivity and signal-to-noise ratio of the telemetry microphone. The reconstructed acoustic scenes offer an avenue for detailed analysis of important cues for figure-ground separation in a cluttered environment, and serve as a basis for subsequent neurocomputational modeling of auditory scene analysis performed by the bat's sonar receiver.

Session 1pAO

Acoustical Oceanography and Signal Processing in Acoustics: Using Acoustics to Study Fish Distribution and Behavior II

Kelly J. Benoit-Bird, Cochair
College of Earth, Ocean & Atmos. Sci., Oregon State Univ., 104 CEOAS Admin Bldg., Corvallis, OR 97331

Timothy K. Stanton, Cochair
Woods Hole Oceanogr. Inst., MS #11, Woods Hole, MA 02543-1053

Contributed Papers

1:00

1pAO1. Concurrent inversion of bio and geoacoustic parameters from broadband transmission loss measurements in the Santa Barbara Channel. Orest Diachok and Glenn Wadsworth (Johns Hopkins Univ. APL, 11100 Johns Hopkins Rd., Laurel, MD 20723, orestdia@aol.com)

This paper describes result of an interdisciplinary experiment, BAS II, which included coincident measurements of broadband (0.3–5 kHz) transmission loss (TL), depths and length distributions of fish, geoacoustic parameters, and continuous temperature profiles. The objective: demonstrate the accuracy of bioacoustic parameters of fish inverted from TL data. TL measurements were conducted between a fixed source and a fixed 16 element, receiving array that spanned most of the water column. Trawls revealed that the dominant species were sardines and anchovies. The TL data at night exhibited absorption lines at resonance frequencies associated with 15 and 8 cm long sardines, and 11, 9.5, and 5.5 cm long anchovies at 12 m, in good agreement with coincident trawl and echo sounder measurements. TL data during the day exhibited an absorption line associated with sardine schools. Concurrent inversion of bio and geoacoustic parameters of nighttime data was based on the Genetic Algorithm. The layer of fish was characterized by selectable depth, thickness and attenuation coefficient. Inverted values of biological parameters at 1.1 kHz, the resonance frequency of 11 cm sardines, were in accord with echo sounder and trawl data.

1:15

1pAO2. Spatiotemporal variability of clutter due to fish aggregations in the vicinity of Heceta Bank. Roger C. Gauss, Joseph M. Fialkowski (Acoust. Div., Naval Res. Lab., Code 7164, 4555 Overlook Ave., S.W., Washington, DC 20375-5350, roger.gauss@nrl.navy.mil), and Richard H. Love (BayouAcoust., Abita Springs, LA)

Acoustic interactions with fish can be a significant source of clutter to mid-frequency (MF; 1–10 kHz) active sonars. To develop robust schemes for reducing sonar false alarm rates, it is thus important to accurately characterize the spatiotemporal nature of MF echoes from fish. In the summer of 2012, long-range MF measurements of coherent backscattering from fish aggregations were made at several sites in the vicinity of Heceta Bank off the coast of Oregon. The dominant fish species included Pacific sardines that typically school near the surface and Pacific hake that typically are above the bottom in loose aggregations or layers, and so represent spatially different echo classes. Measured spatiotemporal fish echo statistics are discussed in terms of the observed in-situ distribution and behavior of these two species. NRL's moment-based Poisson-Rayleigh clutter model is used to physically relate the measured amplitude distributions to scatterer attributes (spatial density and dispersion). The temporal persistence of the clutter echoes is quantified via georeferenced echoes over multiple broadband transmissions. The results suggest that in this area, while fish may present a significant source of false alarms, their impact can be reduced by exploiting their short time-scale behavior. [Work supported by the Office of Naval Research.]

1:30

1pAO3. Accounting for the non-Rayleigh echo statistics of individual elongated scatterers in an aggregation. Wu-Jung Lee (Inst. for Systems Res., Univ. of Maryland, 1147 Biology/Psych. Bldg., College Park, MD 20742, wjlee@umd.edu) and Timothy K. Stanton (Dept. of Appl. Ocean Phys. and Eng., Woods Hole Oceanographic Inst., Woods Hole, MA)

The statistics of echoes from active sonar systems, such as the shape of the probability density function (pdf) of echo magnitude, can be used to estimate the numerical density of scatterers in aggregations. This study investigates the importance of non-Rayleigh echo statistics of individual scatterers on the shape of the echo pdf of aggregations in which the echoes from individuals may overlap. The signals are broadband, and the geometry involves direct paths between the sonar and the scatterers without interference from the boundaries. Echo pdf models are generated by varying the number of scatterers randomly distributed in a half-space shell while accounting for the frequency-dependent system response and beampattern effects. Individual scatterers are modeled using elongated spheroidal shapes with varying distributions of lengths and angles of orientation. The non-Rayleigh echo statistics of individual scatterers were found to contribute significantly to the non-Rayleigh characteristics of the echo pdf of aggregations of those individuals. This model is applied to estimate the numerical density of fish in aggregations observed using broadband signals (30–70 kHz) in the ocean. The results show the importance of incorporating realistic parameters for modeling individual scatterers in echo statistical analyses.

1:45

1pAO4. Acoustic scattering characteristics of pelagic and coastal nekton. Joseph Warren, Kaylyn Becker (Stony Brook Univ., 239 Montauk Hwy, Southampton, NY 11968, joe.warren@stonybrook.edu), Dezhang Chu (NWFSC, NOAA, Seattle, WA), and Kelly Benoit-Bird (COAS, Oregon State Univ., Corvalis, OR)

Measurements of several acoustic scattering characteristics were made for a variety of different nekton. At-sea broadband high-frequency (100s to 1000s kHz) backscatter measurements were made on several species of myctophids and other types of pelagic nekton (e.g. fish, shrimp). Animals were caught in mid-water trawls off the coast of Oregon during the summer of 2012, and measurements were made on fresh specimens from multiple species. Both broadside (dorsal) and end-on (head/tail) measurements were recorded. There was strong variability among and within species as well as with animal orientation. We also report measurements of swim-bladder size, shape, and fullness for the pelagic myctophids. Additionally, high-resolution computerized tomography (CT) scans were made for several coastal nekton (including squid, silverside, and sea bass) species from New York. These scans provide information on the density contrast of the various organs and other structures in the animal. These data provide useful information for acoustic scattering models of these and other similar animals.

2:00

1pAO5. Modeling the acoustic color of large aggregations of fish. David Burnett (Naval Surface Warfare Ctr., Code CD10, 110 Vernon Ave., Panama City, FL 32407, david.s.burnett@navy.mil)

The Naval Surface Warfare Center Panama City Division has developed a 3-D finite-element computer simulation system, PC-ACOLOR, for modeling the "acoustic color" (target strength as a function of frequency and aspect angle) of realistic elastic objects, either singly or in aggregations, that are near the bottom of the ocean or in deep water. It employs 3-D continuum mechanics throughout the entire computational domain. All objects and fluids are treated collectively as a single heterogeneous continuum; no engineering approximations are used anywhere. PC-ACOLOR was developed originally for modeling manmade structures, but it is intrinsic to finite-element modeling that the same code can be applied, unaltered, to any elastic objects, whether they be manmade structures or biological organisms. The talk will give an overview of PC-ACOLOR: structural acoustic concepts; modeling techniques; verification and validation; and examples of different types of objects that have been modeled, including aggregations of many fish.

2:15

1pAO6. Material properties of Pacific hake, Humboldt squid, and two species of myctophids in the California Current. Kaylyn Becker and Joseph D. Warren (School of Marine and Atmospheric Sci., Stony Brook Univ., 239 Montauk Hwy, Southampton, NY 11968, kaylyn.becker@gmail.com)

We measured the material properties of Pacific hake (*Merluccius productus*), Humboldt squid (*Dosidicus gigas*), and two species of myctophids (*Symbolophorus californiensis* and *Diaphus theta*) collected from the California Current. Density contrast (g) was measured for pieces of hake and myctophid flesh, and the following Humboldt squid body parts: mantle, arms, tentacle, braincase, eyes, pen, and beak. Density contrast varied with fish species, as well as among squid body parts. Effects of animal length and environmental conditions on nekton density contrast were investigated. Sound speed contrast (h) was measured for hake and myctophid flesh, Humboldt squid mantle and braincase, and varied within and between nekton taxa. These material property measurements can be used to more accurately parameterize target strength models and increase the accuracy of nekton biomass from acoustic surveys.

2:30

1pAO7. Target phase information for acoustic target identification: Method and preliminary results. Alan Islas-Cital (R&D, Navico, 4500 S. 129th East Ave. Ste. 200, Tulsa, OK 74134-5885, alan.islas@navico.com), Rubén Picó (Departamento de Física Aplicada, Universitat Politécnica de Valéncia, Gandia, Spain), and Phil Atkins (School of Electron., Elec. and Comput. Eng., Univ. of Birmingham, Birmingham, United Kingdom)

Target phase information in acoustic backscattering has been proposed as an additional target identification parameter. In general, incorporating phase into sonar signal processing for acoustical oceanography could aid in the assessment of fish populations and ecosystems. In this work, a broadband sonar system calibrated in amplitude and phase is used to measure the response of submerged targets in a laboratory water tank. Frequency domain data processing is applied, with target phase measured as a phase angle difference between two frequency components. This approach aims to eliminate range factors, leaving only target-induced phase features. The method is developed and validated by comparing experimental results to analytical and numerical methods, in the characterization of some targets with regular geometries such as spheres, shells, and cylinders. A compensation algorithm is implemented to account for phase ambiguities and arrive to a figure of merit for template classification. Simplified scenarios are studied in order to demonstrate the potential applicability of this method.

2:45–3:00 Break

3:00

1pAO8. Calibration of a broadband acoustic system in near-field. Grant Eastland (NW Fisheries Sci. Ctr., Frank Orth & Assoc. (NOAA Affiliate), 2725 Montlake Blvd. E, F/NWC4, Seattle, WA 98112-2097, grant.eastland@noaa.gov) and Dezhang Chu (NW Fisheries Sci. Ctr., NOAA Fisheries, Seattle, WA)

This paper investigates the applicability of calibrating a broadband acoustic system in Near-field. The calibration was performed on a single transducer with a monostatic or backscattering configuration using a standard target, a 25-mm tungsten carbide sphere, in the near-field of both the transducer and the sphere. Theoretical model to quantify the near-field effect was developed in the paper. Theoretical simulations revealed that although the shape of the frequency responses of the received echoes at different distances varied significantly, the null positions were essentially invariant, a unique characteristic that was used to determine the compressional and shear wave speeds in the calibration sphere. The calibration curves obtained at different distances in the near-field by taking into account the near-field effect were consistent with each other. Since the transducer was located in the near-field, the signal-to-noise ratio was high, resulting in a much wider usable bandwidth, between 300 and 800 kHz, than the nominal bandwidth. The methods reported here could potentially be applied to the calibration of multibeam echosounder and sonar systems.

3:15

1pAO9. Model-based and *in-situ* observations of high-frequency (10s–100s kHz) acoustic scattering from multiple targets. Samuel S. Urmy and Joseph D. Warren (Marine and Atmospheric Sci., Stony Brook Univ., 239 Montauk Hwy., Southampton, NY 11968, samuel.urmy@stonybrook.edu)

The biomass of many fish and plankton stocks is estimated using active acoustics and the echo-integration method. This method relies on the assumption that the acoustic energy backscattered by schools or aggregations varies linearly with the number of scattering targets. While accurate under most circumstances, theory predicts this assumption will break down at high scatterer densities, a prediction that has been confirmed experimentally in previous studies. The number of studies exploring these effects, however, remains small. Departures from linearity may be caused by multiple scattering, shadowing effects, and/or resonant spacing of targets at the ensonifying frequency. We explored these effects on different configurations of scatterers using a combination of theory, computer modeling, and *in-situ* measurements on standard targets. The computer model in particular facilitated controlled testing on scattering configurations that would be difficult to achieve *in-situ*. We found that the linearity assumption is supported at most densities commonly encountered in fisheries surveys, but that there are some situations where nonlinear effects become important and should be considered. These results will be useful in the interpretation of echo-integration data from a variety of species and ecosystems.

3:30

1pAO10. Comparison of near-field acoustic coherent backscattering simulations with optics theory and experiments. Adaleena Mookerjee and David R. Dowling (Mech. Eng., Univ. of Michigan, 2010 W.E. Lay Automotive Lab., 1231 Beal Ave., Ann Arbor, MI 48109, adaleena@umich.edu)

Remote discrimination of fish schools from other scatterers in the water column is important for environmental awareness and monitoring, and for a variety of sonar applications. Depending on the school's geometry and number of fish, and the fish's scattering characteristics, there may be preferential backscatter of sound, a phenomenon known as coherent backscatter enhancement (CBE). Thus, the presence and characteristics of CBE could help in discriminating fish schools from other objects. For CBE, the addition of in-phase scattered waves from the propagation path pairs yields a scattered intensity enhancement of a factor of two in the direction opposite to that of the incident wave. However, prior simulations based on the Foldy (1945) equations have suggested the enhancement may be greater than a factor of two for relatively large closely spaced scatterers. This presentation re-examines this topic and provides new CBE simulation results for near field scattering from finite-sized aggregations of point scatterers. Comparisons are made with equivalent results from the optics experiments of Wolf and Maret (1985) and the theory of Akkermans *et al*. (1986). Extension of this effort to comparisons of CBE from single frequency and broadband pulse illumination is anticipated. [Sponsored by the Office of Naval Research.]

3:45

1pAO11. Time domain investigations of acoustical scattering from schools of swim bladder fish. Maria P. Raveau (Departamento de Ingenieria Hidraulica y Ambiental, Pontificia Universidad Catolica de Chile, Vicuña Mackenna 4860, Macul, Santiago 7820436, Chile, mpraveau@uc.cl) and Christopher Feuillade (Departamento de Fisica, Pontificia Universidad Catolica de Chile, Santiago, Chile)

Recent studies of time independent scattering from schools of swim bladder fish reveal important differences between the predictions of a mathematical model, which fully incorporates multiple scattering processes between the fish, and a second approach which treats the school as an effective medium with complex sound speed determined by the swim bladder resonance function. In back scattering, both modeling and data comparisons show that the effective medium approach underestimates the scattering amplitude when the fish separation is greater than about a quarter of the incident wavelength. In contrast, comparisons in the forward scattering direction show good agreement. These results are critically significant for time domain investigations of fish school scattering, aimed at using spatial and temporal variations in the acoustic field to study the stochastic behavior of the distribution and motion of the fish ensembles. A simple approach, using the inverse FFT of the school model and effective medium harmonic solutions, again reveals the limitations of the effective medium approach in back scattering. However, to obtain high resolution time sampling, more sophisticated time domain solution techniques based upon numerical integration and perturbation theory approaches are necessary. Both computational studies and data comparisons will be presented. [Research supported by ONR.]

4:00

1pAO12. Extracting effective medium properties for fish schools from resonator and free-field measurements. Craig N. Dolder, Gregory R. Enenstein (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, dolder@utexas.edu), Kevin M. Lee (Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, Austin, TX)

Resonator and free-field measurements were performed with both real and model fish schools in order to determine the relationship between fish school density and sound speed and absorption of acoustic waves. The results are compared to effective medium models for sound propagation through fish and encapsulated bubbles. The current species under study is *Danio rerio* (zebrafish); however, this technique can be extended to other fish species. [Work supported by ONR.]

4:15

1pAO13. Exploration of a Bloch wave expansion technique for analyzing backscattering from large fish schools. Jason A. Kulpe, Michael J. Leamy, and Karim G. Sabra (Georgia Inst. of Technol., 771 Ferst Dr., Rm. 136, Atlanta, GA 30332, jkulpe@gatech.edu)

Scattering from large fish schools at typical SONAR frequencies (1–10 kHz) can significantly contribute to volume reverberation. An efficient modeling technique, which accounts for fish' spatial configuration and inter-fish multiple scattering effects, is sought to quantify the acoustic scattering from a large fish school. Recent work has exploited the near-periodic nature of the schooling fish to represent the school as an equivalent phononic crystal (PC) composed of air-filled swim bladders periodically arrayed in a water matrix. Using the Bloch theorem applied to infinite media discretized by finite elements, the approach is capable of quickly and accurately generating the band structure and reflection/transmission coefficients for an incident plane wave. In this work, we extend the analysis approach to finite-sized fish schools to quantify the backscattering as a function of incident wave frequency, school geometry, and weak internal disorder. Scattered field predictions are compared against a self-consistent scattering method as well as full-field finite element simulations. For the presented approach, we note fast and scalable computation (with respect to frequency and school size) with very good agreement in predicted scattered pressure fields and the frequencies corresponding to peak target strengths. This work shows promise for predictive SONAR modeling.

4:30–4:55 Panel Discussion

Session 1pBA

Biomedical Acoustics: Breast Ultrasound II

Koen W. A. van Dongen, Cochair
Lab. of Acoust. Wavefield Imaging, Faculty of Appl. Sci., Delft Univ. of Technol., P.O. Box 5046, Delft 2600 GA, Netherlands

Timothy E. Doyle, Cochair
Physics, Utah Valley Univ., MS 179, 800 W. University Parkway, Orem, UT 84058-5999

Invited Papers

1:00

1pBA1. Optimization of the aperture and transducers of a three-dimensional ultrasound computer tomography system. Hartmut Gemmeke, Michael Zapf, Torsten Hopp, and Nicole V. Ruiter (Inst. for Data Processing and Electronics, Karlsruhe Inst. of Technol., Postfach 3640, Karlsruhe 76021, Germany, hartmut.gemmeke@kit.edu)

In previous work we optimized the aperture of our 3D Ultrasound Computer Tomography (USCT) system with emphasis on reflection tomography. Based on the promising clinical results with speed of sound and attenuation images, the next generation aperture will be upgraded to contain also optimization for transmission tomography. The main changes to be implemented in aperture, transducers, and transducer arrays are: (1) Overall 3D aperture: due to the shape of the buoyant breast a simpler hemispherical aperture can be applied. (2) The diameter of the aperture will be increased and the diameter of the transducers will be decreased for further homogenization of the illumination. (3) The disjunctive sampling of the transducers will be increased and the transducers will be distributed randomly to enhance the uniformity of transmission tomography. (4) Transducers will be connected both as emitters and receivers to decrease the need for mechanical movement of the aperture.

1:20

1pBA2. A practical, robust approach to high resolution ultrasonic breast tomography. Peter Huthwaite (Mech. Eng., Imperial College London, City and Guilds Bldg., Exhibition Rd., London SW7 4JU, United Kingdom, p.huthwaite@imperial.ac.uk) and Francesco Simonetti (School of Aerosp. Systems, Univ. of Cincinnati, Cincinnati, OH)

Breast ultrasound tomography is considered a potentially safer, more reliable, and more sensitive alternative to the widely used mammography for breast cancer diagnosis and screening. Vital to achieving this potential is the development of imaging algorithms to unravel the complex anatomy of the breast. Bent Ray Tomography (BRT) is the most prominent algorithm, producing sound-speed maps, but the underlying approximation of ray theory means that the algorithm is unsuitable for structures where significant diffraction is present. Accordingly, the maximum resolution of the BRT algorithm is not sufficient to image the details of the breast on the scale of a few millimeters. Therefore, iterative full-wave inversion techniques are often applied to improve the resolution of the BRT image, but they are typically slow or fail because of the uncertainties such as 3D effects, noise, or transducer characteristics. Presented here is a solution where the BRT algorithm is combined with diffraction tomography (DT), avoiding iterations yet producing a high resolution sound-speed image. It is demonstrated with both numerical and experimental data how this can successfully—and robustly—deal with a range of phenomena present in breast ultrasound experiments such as attenuation, density, 3D effects, and transducer directivity while maintaining a high resolution.

1:40

1pBA3. Using higher-order scattering in seismic imaging. Dirk J. Verschuur (Faculty of Appl. Sci., Delft Univ. of Technol., Lorentzweg 1, Delft 2628 CJ, Netherlands, d.j.verschuur@tudelft.nl)

For seismic exploration, acoustic sources and receivers are positioned at the earth's surface in order to measure the reflection response from the subsurface inhomogeneities. However, in most current imaging algorithms, only the primary reflections are being taken into account and multiple reflections are being discarded as noise. In the recently proposed method of full wavefield imaging, the higher-order scattering is taken into account in the imaging process where they contribute to extend the illumination area and no longer produce spurious imaging artifacts. This method involves an inversion process, where a recursive modeling method is used to predict the measured reflection response—including all its higher-order scattering effects—based on estimated reflectivity values and background velocity model. Because the background velocity cannot be assumed to be homogeneous in the earth, background velocity model estimation is a crucial component of the imaging process. It appears that automatic estimation of velocity models can be accomplished within the concept of full wavefield migration, in which higher-order scattering events can also be fully accommodated. Finally, the results of full wavefield imaging can be combined with localized, target-oriented full waveform inversion in order to get the final details, being the elastic properties, at the target zone.

2:00

1pBA4. On the separate recovery of spatial fluctuations in compressibility and mass density in pulse-echo ultrasound imaging using linear inverse scattering. Martin F. Schiffner and Georg Schmitz (Medical Eng., Ruhr-Univ. Bochum, Universitätsstr. 150, Bochum 44801, Germany, martin.schiffner@rub.de)

In pulse-echo ultrasound imaging (PEUI) of soft tissues, the scattered sound field is governed by spatial fluctuations of the two mechanical parameters compressibility and mass density. Spatial fluctuations in compressibility act as isotropic monopole radiators while spatial fluctuations in mass density act as anisotropic dipole radiators. Conventional strategies for linear image reconstruction in PEUI, e.g., delay-and-sum, minimum variance, and synthetic aperture focusing, exclusively account for monopole scattering. This neglect of the inhomogeneous mass density might be accompanied by a loss of diagnostically relevant information, e.g., the detection of tissue abnormalities. In this study, we formulate a linear inverse scattering problem to recover separate, space-resolved maps of the spatial fluctuations in both mechanical parameters from measurements of the scattered acoustic pressure. The physical model accounts for frequency-dependent absorption and dispersion in accordance with the time causal model. The computational costs are effectively reduced by the usage of the fast multipole algorithm. The concept is evaluated using simulated and experimentally obtained radio frequency data.

2:20

1pBA5. A closer look at contrast source inversion for breast cancer detection. Koen W. A. van Dongen and Neslihan Ozmen-Eryilmaz (Lab. of Acoust. Wavefield Imaging, Faculty of Appl. Sci., Delft Univ. of Technol., P.O. Box 5046, Delft 2600 GA, Netherlands, k.w.a.vandongen@tudelft.nl)

Ultrasound is an emerging technology for breast cancer detection. It is an efficient and harmless method that can detect tumors in dense breasts, which may be missed using mammography. Currently, several fully automated ultrasound screening modalities are being developed. For some of those systems, accurate knowledge about the transducer location is available, making the measured data suitable for imaging using non-linear inversion methods. A promising, but costly, inversion method is contrast source inversion, which has been tested successfully on synthetic measurement data. To reduce the computational costs (computing time and memory load) various setups were tested. Results obtained with synthetic data show that inversion using a single frequency component only, still yielded excellent imaging results, while significantly reducing memory requirements. In addition, our data indicate that the number of source positions is less important than the number of receiver positions. Thus, while keeping the total number of A-scans identical, the inversion improved when the number of source positions was reduced and the number of receiver positions was increased. This approach efficiently reduced the computational costs associated with the inversion.

2:40–3:00 Break

Contributed Papers

3:00

1pBA6. Reproducibility of high-frequency ultrasonic signals in breast cancer detection. A. Mackay Breivik, Andrew J. Marshall (Biology, Utah Valley Univ., 800 W. University Parkway, Orem, UT 84058-5999, mackay.breivik@gmail.com), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

The central question of this project was to determine the reproducibility of high-frequency (HF) ultrasonic signals in breast cancer detection. Previous studies on surgical specimens of breast tissue have shown that HF ultrasound (20–80 MHz) appears sensitive to a range of breast pathologies. A measurement in the ultrasonic signal called the peak density appears most sensitive to the pathology of the breast. The reproducibility of this parameter has not been quantitatively measured in a comprehensive manner. In parallel to a clinical study being conducted at the Huntsman Cancer Institute, the reproducibility of peak density measurements was studied using chicken and bovine tissue. Tissue was cut into $4 \times 3 \times 0.5$ cm and $4 \times 3 \times 1.5$ cm cubes and tested at 23.4°C. Waveforms were obtained for two types of measurements: (1) where the transducer stayed in contact with the tissue, and (2) where the transducer was lifted from the tissue between measurements. Spectral peak densities were obtained from 640 measurements. Type 1 measurements showed high reproducibility. Type 2 measurements displayed greater variability but were consistent with previous measurements on lumpectomy tissue specimens. The variability of the type 2 measurements is believed to be due to transducer-induced pressure differences between each measurement and is currently being studied.

3:15

1pBA7. Utah Valley University/Huntsman Cancer Institute Collaborative Breast Cancer Study: High-frequency ultrasound for margin assessments. J. Andrew Chappell, Janeese E. Stiles (Biology, Utah Valley Univ., Orem, UT), Leigh A. Neumayer (Surgery, Univ. of Utah, Salt Lake City, UT), Rachel E. Factor (Pathol., Univ. of Utah, Salt Lake City, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., MS 179, 800 W. University Parkway, Orem, UT 84058-5999, Timothy.Doyle@uvu.edu)

In a joint effort between Utah Valley University and the Huntsman Cancer Institute, high-frequency (HF) ultrasound (20–80 MHz) is being studied to determine the pathology of surgical margins from breast conservation surgery. Results from a 2010 NIH R21 study indicated that multiple parameters in the HF ultrasonic spectrum correlate to a range of breast tissue pathologies. This technology promises to provide rapid, intraoperative evaluation of surgical margins, thereby decreasing the number of additional surgeries for patients. A blind study is currently being conducted with conventional pathology as the gold standard for assessing the accuracy of the method. Specimens are delivered by the surgeon's team immediately following resection and ultrasonically tested outside the surgical suite. The margins are approximately $3 \times 20 \times 20$ mm and are oriented using a small staple inserted by the surgeon in one corner and a stitch on one side. The margin is tested at 2–5 locations and then sent to pathology for analysis. Pathology and HF ultrasound results will be compared for correlation at the end of the study, which is expected to last one year. The study will include approximately 80 patients, 360 tissue samples, and 1400 tested locations. If successful, the method will move into clinical trial.

3:30

1pBA8. High-frequency ultrasound study of tissue margins from breast conservation surgery: Preliminary results. Teresa L. Wilson, Amy A. Fairbrother (Biology, Utah Valley Univ., 800 W. University Parkway, Orem, UT 84058-5999, tlwilson59@gmail.com), Monica Cervantes, and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

A critical issue in breast conservation surgery (lumpectomy) for breast cancer treatment is ensuring the tissue surrounding the excised tumor, the margins, are cancer-free. In collaboration with the Huntsman Cancer Institute at the University of Utah, researchers from Utah Valley University are using high-frequency (HF) ultrasound to test the pathology of lumpectomy surgical margins. This preclinical study is a blind study which will involve 80 patients, approximately 320 specimens, and use traditional pathology as the "gold standard" for measuring the accuracy of the HF ultrasound method. Ultrasonic waveforms of margins were acquired at the Huntsman Cancer Hospital in pitch-catch and pulse-echo modes using 50-MHz transducers with 6.35 mm-diameter active elements. The data were analyzed to obtain ultrasonic parameters such as ultrasonic wavespeed, attenuation, and spectral peak density (the number of peaks and valleys in a HF ultrasonic spectral band). Preliminary results indicate variations in peak density between margin specimens and individual locations on specimens that are indicative of malignant and atypical breast pathologies. The objective of this paper is to search for trends in the data acquired to date to provide an assessment of reliability, stability, and robustness of the study.

3:45

1pBA9. Ultrasonic phenotyping of breast cancer cells for molecular subtyping. Laurel A. Thompson (Chemistry, Utah Valley Univ., 800 W. University Parkway, Orem, UT 84058-5999, laurelathompson@gmail.com), Janeese E. Stiles, J. Andrew Chappell, Ashley N. Calder, Caitlin Carter, Janice E. Sugiyama (Biology, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Breast cancer can be divided into molecular subtypes which are defined by their genetic and protein expression profiles. Current methods aimed at testing these biochemical signatures are effective classifiers but are not easily transferable to real-time clinical applications. The rapid, cost-effective determination of molecular subtype by other means would be a significant advancement in cancer detection and treatment. Our studies suggest that high-frequency ultrasound (10–100 MHz) may be sensitive to variations among breast cancer subtypes through their cytoskeletal structures, which have distinct biomechanical signatures. To further test this hypothesis, four breast cancer cell lines of different subtypes were cultured and ultrasonically tested. Direct pulse-echo measurements were collected from the cell layers using a 50-MHz transducer immersed in the growth media of the culture plates. Cell reflections in the waveforms were isolated and spectrally analyzed using computationally modeled spectra and principal component analysis (PCA). Cell phenotypes were profiled by using heat maps to display the relative distances between the PCA scores of the experimental and model spectra. The results indicate the phenotype and thus molecular subtype of cancer cells could potentially be determined by comparing their measured spectra to model spectra using a feature classification program such as PCA.

4:00

1pBA10. High-frequency ultrasonic measurement of vascularization in phantoms and Avastin-treated mice with breast tumors. Andrea N. Quiroz, Michaelle A. Cadet (Biology, Utah Valley Univ., 800 W. University Parkway, Orem, UT 84058-5999, aquiroz1912@gmail.com), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Tissue vascularization is an important aspect of tissue engineering and oncology. The objective of this study was to determine if direct ultrasonic measurements in the 10–100 MHz range could be used as an *in vivo* vascularization assay. To simulate vascularization of tissue, phantoms were fabricated from agarose gel inclusions embedded in a gelatin-soluble fiber mixture. Ultrasonic tests were performed using two broadband ultrasonic transducers centered at 50 MHz. Results showed the samples with multiple agarose vasculature structures decreased the ultrasound wavespeed. As the level of vasculature decreased, the wavespeed of the ultrasound increased. Further investigation of vascularization included the *in vivo* evaluation of grafted breast cancer tumors in mice. The experimental group was composed of mice treated with Avastin, an angiogenesis inhibitor. The heterogeneity of the vasculature in the control tissue resulted in the scattering of the ultrasound, decreasing the wavespeed. Because the treated group contained less vascularized, more homogeneous tissue, the wavespeed was significantly higher. Results from both the phantom and mouse tumor studies revealed that the ultrasound wavespeed was inversely proportional to the level of vasculature. The results indicate that direct ultrasound wavespeed measurements in the 10–100 MHz range can be used to identify vascularization in tissue.

Session 1pID

Interdisciplinary Student Council: Introduction to Technical Committee Research and Activities: Especially for Students and First-Time Meeting Attendees

Whitney L. Coyle, Cochair
The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802

Matthew D. Shaw, Cochair
Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802

Chair's Introduction—2:00

Invited Papers

2:05

1pID1. An Introduction to the Acoustical Oceanography Technical Committee. Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu)

The Acoustical Oceanography (AO) technical committee focuses on the development and use of acoustical techniques to understand the physical, biological, geological, and chemical parameters and processes that occur in the ocean interior and its boundaries. Acoustical techniques are uniquely suited for investigating underwater environments due to the low attenuation of sound relative to other commonly used remote sensing techniques based on electromagnetic radiation. The research encompassed by the AO technical committee is highly inter- and multi-disciplinary, with strong overlap with research addressed in the Animal Bioacoustics, Signal Processing, and Underwater Acoustics technical committees. In this presentation, an overview of recent "hot topics" in Acoustical Oceanography will be given, in addition to embarking on a discussion of the role of the AO technical committee in fostering education and leading the way in determining new and innovative directions in the specialization area.

2:15

1pID2. Highlights of the Underwater Acoustics Technical Committee at the 167th Meeting of the Acoustical Society of America. Marcia Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

The underwater acoustics (UW) technical committee (TC) is a dynamic group of investigators researching such varied fields as acoustic tomography, underwater acoustic communications, and shallow water waveguide propagation. In this overview of the UW TC, a brief introduction of the goals and interests of the community will be followed with a series of highlights of what one can expect at the UW sessions.

2:25

1pID3. An introduction to the Physical Acoustics Technical Committee activities. Steven L. Garrett (Grad. Prog. in Acoust., Penn State, Appl. Res. Lab., P. O. Box 30, State College, PA 16804-0030, sxg185@psu.edu)

The primary activity of any ASA Technical Committee is to use collective wisdom of the Committee's membership to determine which research topics within its specialization area are most active and interesting. Based on that assessment, the Committee organizes special sessions at future meetings that will bring together experts from those areas, not necessarily limited to the Society members, who can share interesting results and provide guidance regarding the directions that will lead to further understanding. In Physical Acoustics, that is a particularly daunting challenge given the scope of topics that fall within its purview: use of sound to probe material properties, sound propagation and attenuation mechanisms on this planet and in other parts of the universe, and physical effects of sound and its interaction with other forms of radiation, all of which could also go well beyond the limitations of a linear acoustical theory. Needless to say, involvement in debates about "what's hot" is both interesting and educational. Other activities include proposals for Technical Initiatives that allocate ASA resources. Recently, PATC received funding to co-sponsor the Physical Acoustics Summer School.

2:35

1pID4. Introduction to the Structural Acoustics and Vibration Technical Committee. James E. Phillips (Wilson, Ihrig & Assoc., Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

The Structural Acoustics & Vibration Technical Committee (SAVTC) includes the study of motions and interactions of mechanical systems with their environments and the methods of their measurement, analysis, and control. This talk will provide a broad overview of the many research areas of interest to SAVTC. A few topics will be explored in more depth to provide background on some of the more common analysis methods used by members of the technical committee.

2:45

1pID5. Noise and its impact on our world. Erica Ryherd (Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA 30332-0405, erica.ryherd@me.gatech.edu)

Noise invades all aspects of our lives. The word noise is actually derived from the Latin word "nausea," with one possible connection being that unpleasant sounds were made by seasick passengers or sailors in ancient times. In modern times, the demand for noise research and consulting has intensified in concert with rising population densities, growing industrialized societies, escalating demands from consumers, and increasingly common standards and legislation related to noise. The Acoustical Society of America Technical Committee on Noise (TC Noise) is concerned with all aspects of noise, ranging from noise generation and propagation, to active and passive methods of controlling noise, to the effects of noise on humans and animals. This talk will explore the broad topic of noise and its impact on our world.

2:55

1pID6. Architectural Acoustics-Space for sound, and you. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

The discipline of Architectural Acoustics consistently produces more than 100 papers across six or more special sessions, at each meeting of the ASA. Student paper awards, student design competitions, and Knudsen lectures augment these activities. Joint sessions, particularly with Noise, Musical Acoustics, Psychological and Physiological Acoustics, and Signal Processing in Acoustics, add more still to the architectural acoustics goings-on at every ASA conference. The sphere of influence is not limited to ASA alone, as TCAA members participate in the Green Construction Code of the International Code Council, Society of Motion Picture and Television Engineers Study Group: Movie Theater Sound System Measurement and Adjustment Techniques, Classroom Acoustics Standards, the American Institute of Architects Continuing Education System, and more. This busy committee also produces a steady stream of publications documenting recent work and deciphering standards for key stakeholders. Anyone with an interest in the field will find many opportunities to advance their own expertise, build a network of colleagues, friends, and mentors, and contribute to the essential activities of the Technical Committee on Architectural Acoustics.

3:05–3:20 Break

3:20

1pID7. Overview of Signal Processing in Acoustics. Richard L. Culver (Appl. Res. Lab., Penn State Univ., Po Box 30, 16804, State College, PA 16801, r.lee.culver@gmail.com)

The Signal Processing Technical Committee (SPTC) of the ASA provides a forum for discussion of signal processing techniques that transcend one acoustic application. Signal processing research typically presented at ASA meetings includes techniques that show promise in one application—say underwater acoustics—but may also have application to other areas, for example, speech processing or room acoustics. There are several good reasons to get involved in the SP TC. First, since signal processing is an important aspect of many acoustic research areas, you will have the opportunity to better understand new and potentially useful tools. Second, Signal Processing is a small technical committee and you can make an immediate contribution. This talk provides an overview of some of the current topics in Signal Processing.

3:30

1pID8. The Engineering Acoustics Technical Committee—Where practical applications begin. Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ., N-249 Millennium Sci. Complex, University Park, PA 16803, sct12@psu.edu)

Engineering Acoustics encompasses the theory and practice of creating tools for investigating acoustical phenomena and applying knowledge of acoustics to practical utility. This includes the design and modeling of acoustical and vibrational transducers, arrays, and transduction systems in all media and frequency ranges; instrumentation, metrology, and calibration; measurement and computational techniques as they relate to acoustical phenomena and their utility; and the engineering of materials and devices.

3:40

1pID9. An Introduction to the Musical Acoustics Technical Committee. Paul A. Wheeler (Elec. and Comput. Eng., Utah State Univ., 1595 N 1600 E, Logan, UT 84341, paul.wheeler@usu.edu) and Andrew C. Morrison (Joliet Junior College, Joliet, IL)

The technical committee on musical acoustics (TCMU) is focused on the discovery of novel advancements in the science and technology of music and musical instruments. Many of our members pursue topics related to the physics of musical sound production, music perception and cognition, and the analysis and synthesis of musical sounds and composition. The TCMU draws from many fields represented in the society. Our sessions have presentations made by scientists, engineers, musicians, instrument builders, psychologists, architects, and others interested in the study of the science of music. An overview of selected research topics and activities of the TCMU will be presented.

3:50

1pID10. Overview of Speech Communication research. Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Calhoun Hall 407, 1 University Station B5100, Austin, TX 78712-0198, rajka@mail.utexas.edu)

Even though speech communication is a fundamental part of our daily behavior, its mechanisms are not yet well understood. Speech communication research examines how spoken language is produced, transmitted, and perceived. Speech communication research involves a number of different disciplines, from linguistics and experimental psychology to speech and hearing sciences, and electrical engineering. The field covers a wide range of physiological, psychological, acoustic, and linguistic phenomena. In this talk, I will focus on research examining variation in speech intelligibility, the degree to which spoken language can be comprehended. Even in ideal communicative settings, in quiet environments between normal-hearing, native speakers of a language, speech intelligibility is variable. The variation increases in adverse communication situations that can arise from degradation related to talker (e.g., when second language learners produce non-canonical signal), signal (e.g., when target speech is masked by competing speech), or listener (e.g., when listeners use cochlear implants) characteristics (Mattys *et al*., 2012). Examining speech intelligibility variation from all these perspectives provides insights into the perceptual, physiological, linguistic, cognitive, and neurophysiological mechanisms underlying speech processing. This line of inquiry also has implications for improving speech processing when communicative conditions are compromised.

4:00

1pID11. Psychological and Physiological Acoustics: Investigating the auditory system and its responses to sound. Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov)

Psychological and physiological acoustics concerns the investigation of the auditory system and its responses to sound in humans and other species. This encompasses perception and perceptual organization of simple and complex sounds, including speech; anatomy and function of the auditory pathways, including all physical and biological responses to auditory stimulation; hearing disorders, hearing loss, and auditory prostheses; vibrotactile and vestibular sensation, and the interaction of hearing with other sensory modalities; developmental, aging, learning, and plasticity effects in auditory function; and theories and models of auditory processes. This talk describes several current areas of research, including the benefits of having two ears, how intense noise can damage the auditory nerve, and how computational models of the auditory system can complement behavioral and physiological experiments to broaden our understanding of how the auditory system responds to sound.

4:10

1pID12. Biomedical Acoustics: Making a quantum leap in medicine. Tyrone M. Porter (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, tmp@bu.edu)

At the dawn of a new era in medicine, the future of biomedical acoustics is bright. In the last century, we witnessed the introduction of ultrasound contrast agents, lithotripsy, and the visualization of ultrasound images in three dimensions. Currently, scientists are developing acoustic-based techniques for opening the blood–brain barrier transiently in order to treat brain tumors and neurological diseases. Additionally, researchers are developing echogenic liposomes and microbubbles for targeted ultrasound image enhancement and drug and gene delivery. Further excitement has been generated by the advances made with focused ultrasound to ablate or mechanically erode solid tumors in a noninvasive and site-specific manner. As technology and protocols continue to evolve, biomedical acoustics will have a dramatic impact on the diagnosis and treatment of debilitating diseases, thus improving patient care and quality of life.

4:20

1pID13. Animal Bioacoustics: Sounds and soundscapes. Andrea Simmons (Cognit., Linguistic & Psychol. Sci., Brown Univ., Box 1821, Providence, RI 02912, Andrea_Simmons@brown.edu)

There is considerable biological diversity in the mechanisms and strategies animals use to produce and to perceive acoustic cues and to communicate and navigate within complex soundscapes. Researchers in animal bioacoustics use a variety of experimental techniques, from passive tracking to active training, to understand and model this diversity. I will highlight recent experimental work from a few model species to show how knowledge of environmental acoustics enhances our appreciation of animal evolution and cognition.

Session 1pMU

Musical Acoustics: Topics in Musical Acoustics

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

Contributed Papers

1:30

1pMU1. Turning music into sound: Vincenzo Galilei's contributions to the history of acoustics. Marina Baldissera Pacchetti (History and Philosophy of Sci., Univ. of Pittsburgh, 333 s Pacific Ave., Apt. 3, Pittsburgh, PA 15224, mab360@pitt.edu)

I investigate the contributions of Vincenzo Galilei (1520—1591), father of Galileo Galilei (1564—1642) to the development of acoustic science, with an emphasis on the role of phenomenology of sound and mathematical explanation of consonances. Sixteenth century music theory mainly aimed at recovering standards of ancient Greek music theory, transmitted by the works of Boethius, and later by Claudius Ptolemy's Harmonics (200 AD). Gioseffo Zarlino (1517—1590), a major exponent of Renaissance music theory, relied on a priori mathematical quantification of sound, which was based on a particular class of so-called Pythagorean ratios. Mathematical properties of these ratios were used to justify *a priori* the consonance of contemporary music. This clashed with aesthetic perception of sound. Vincenzo argues in favor of the validity of sense perception: perception and aesthetic judgment are explanatory prior to mathematics, and ratios quantify an (undefined) element of sense perception. To prove his point, Vincenzo presented different experiments, demonstrating that the Pythagorean conception of consonance, according to which the octave was embodied by the ratio 1:2, cannot hold for all sound producing physical systems. The study of Vincenzo's experiments in their historical context provide a novel perspective on the relation between the development of scientific inquiry and perception in the history of acoustics.

1:45

1pMU2. Investigation of acoustical parameters in the South Indian musical intervals. Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882, potty@egr.uri.edu) and Venkitanarayanan Parameswaran (Dept. of Mech. Eng., Indian Inst. of Technol., Kanpur, India)

The South Indian musical style is one of the two prominent styles of classical music styles in India whose roots can be traced back to hundreds of years in the past. The South Indian musical scale is said to have evolved from a set of seven primary notes on the basis of 22 intervals. Present day practice is to use 12 intervals in an octave. A scale is divided in to 12 intervals with fixed values that constitute the basis of musical notes. South Indian classical musical intervals are melodic and are therefore flexible. Compared to the twenty two interval scheme, the 12 interval scheme allows the freedom to perturb the fixed frequency values around their defined positions. In this study, pitch analysis of audio samples is carried out to investigate the deviations from fixed frequency values of musical intervals. Departures from theoretically calculated acoustical values will be demonstrated and discussed. The intervals vary dynamically depending on the artistic individuality and the musical context.

2:00

1pMU3. Clarinet playing frequency predictions: Comparison between analytic and numerical simulations. Whitney L. Coyle (The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, wlc5061@psu.edu), Jean Kergomard, and Philippe Guillemain (LMA-CNRS, Marseille, France)

The input impedance measurement can provide the resonance frequencies of an instrument and is a standard method used by wind instrument makers in designing modifications. For a complete design, it is necessary to know the playing frequencies themselves, which depend on several control parameters, such as the blowing pressure and reed opening and the input impedance. Using the values of these parameters, we can determine the playing frequencies. This research will analytically deduce these frequencies from the different control parameters and from the input impedance curve. Four effects are known to influence the playing frequency and are examined separately: the flow rate due to the reed motion, the reed dynamics, the inharmonicity of the resonator, and the temperature gradient in the clarinet. The results for a particular clarinet are given and compared to numerically simulated playing frequencies. Experimental methods are also presented and discussed.

2:15

1pMU4. A cross correlation of stringed instruments. Katarzyna Pomian (Phys., Loyola Univ. Chicago, 1536 Courtland Dr., Arlington Height, IL 60004, kpomian@luc.edu)

Stringed instruments can be characterized by their individual body shapes, sizes, types of strings, and sounds. Studying the string resonance, body properties, and high speed photography of the strings, it is possible to describe the individual instruments in their specific functions. We gathered data for 13 different instruments and revealed the variances in their sound, based upon the unique characteristics of each instrument. Our focus was on the comprehensive analysis of the each instrument when correlating all the components of the data. We then aligned all the instruments together and investigated how the varying body shapes influence the sound that is produced by the specific instruments. Our analyses showed how these components work together to create the individually characterized sounds of the different instruments. [This paper is complementary to that of Gordon P. Ramsey.]

2:30

1pMU5. A comparative study of stringed instruments. Gordon Ramsey (Phys., Loyola Univ. Chicago, 6460 N Kenmore, Chicago, IL 60626, gprspinphys@yahoo.com)

Many studies for stringed instruments exist. Most analyses have been made on violins, guitars, and pianos, but comparative studies on these and less popular instruments have not been done. Our research consists of an in

depth analysis of a variety of stringed instruments. The experiments included spectrum analysis, body resonances using Chladni patterns, and high-speed videos to visually observe the string oscillation modes. A combination of these methods was performed on 13 different stringed instruments. The spectral analysis was done on all instruments with the strings plucked or picked at different locations. Corresponding high-speed video was taken on many to observe how the waves propagate along the string. The violin and viola were recorded when bowed to compare the images of the stings when plucked. String resonances were compared to the body resonances to see the synthesis between the two. Comparisons of the spectrum, body resonances, and string oscillations have been made between these instruments to gain a better understanding of how they operate and why each emits the unique sounds that it does. [This paper is complementary to the poster of Katarzyna Pomian at this conference.]

2:45–3:00 Break

3:00

1pMU6. Sound power levels of the Caxirola and different types of caxixis. Talita Pozzer and Stephan Paul (DECC-CT-UFSM, UFSM, Undergraduate Program in Acoust. Eng., Tuiuti, 925. Apto 21, Santa Maria, RS 97015661, Brazil, talita.pozzer@eac.ufsm.br)

In 2014, Brazil will host the FIFA World Cup and Brazilian Musician Carlinhos Brown created the caxirola as the official music instrument, adapting an old African instrument—the caxixi. In both instruments, the sound is generated by hard particles impacting on the walls of a closed basket. While the caxirola is made of environmental-friendly polymer, the caxixi is handcrafted of natural components. At a previous ASA meeting [Pozzer and Paul, J. Acoust. Soc. Am. 134(5), 4187 (2013)], we presented both instruments and sound pressure levels measured at users ears. Being handmade makes the caxixi to be highly variable in size and proportions, in contrast to the caxirola which is an industry product. We now present sound power level (SWL) measurements made by the hemi-anechoic room method and the reverberation chamber method. The SWL measured for the caxirola was 79 to 86 dB (80 to 85 dBA) for transversal and longitudinal use, respectively. The SWL of four different caixixis measured ranged between 69 and 80 dB (70 to 79 dBA). Octave band SWLs were found to be higher for the caxirola compared to all four caxixi in the 500 Hz to 4 kHz bands.

3:15

1pMU7. Detection of musical notes and chords using a holistic approach. Arturo Camacho (Comput. Sci. and Informatics, Univ. of Costa Rica, Escuela de Ciencias de la Computación e Informótica, Universidad de Costa Rica, San José, San José 2060, Costa Rica, arturo.camacho@ecci.ucr.ac.cr)

There are currently two main approaches for the automatic recognition of chords: (1) detecting chords from pitch class profiles, which gives information about chroma, but not height, and (2) detecting individual notes, either by peak picking from a score function or by detecting and then canceling individual notes. We propose a new method that combines both approaches: it detects chords in a holistic way, but at the same time, it gives information about the chroma and height of individual notes. The approach consists in computing scores for individual notes and chords, especially those in closed form (i.e., with small intervals between notes), and then picking the candidates with maximum score, either notes or chords. This approach, inspired in the way musicians perceive chords: as a whole and not as individual notes, avoids the iterative approach of detecting and canceling notes. The method works particularly well for notes within small intervals, which tend to be hard to detect in other approaches.

3:30

1pMU8. Perception of combination tones correlated to cochlear activity. Victoria Suha (Elec. and Comput. Eng., Northeastern Univ., #3430 337 Huntington Ave., Boston, MA 02115, victoria.suha@hotmail.com) and Michael Epstein (Speech-Lang. Pathol. and Audiol., Elec. and Comput. Eng., BioEng., Northeastern Univ., Boston, MA)

String players producing double stops at certain intervals perceive a third note that is not physically present (combination tone). This experiment was designed to determine whether or not this combination tone is a result of measurable cochlear activity. Young, normal-hearing musicians matched the pitch and rated the loudness of combination tones perceived in the presence of pairs of pure tones. These matches and ratings were compared with distortion-product otoacoustic emissions (DPOAEs) generated by the same tone pairs. While there was some variability in listener performance, with some listeners making inconsistent pitch matches across trials, listeners who gave consistent responses pitch matched the combination tone to a frequency close to the strongest DPOAE. This suggests that the perception of combination tones is associated with physical activity within the cochlea.

3:45

1pMU9. Investigating acoustic and electroglottograph features to characterize Passaggio in female singers. Shonda Bernadin (Elec. and Comput. Eng., Florida A&M University-Florida State Univ. College of Eng., 2525 Pottsdamer St., Tallahassee, FL 32310, bernadin@eng.fsu.edu), Richard Morris (Commun. Disord., Florida State Univ., Tallahassee, FL), Lance Ellerbe, and Demissew Kessela (Elec. and Comput. Eng., Florida A&M University-Florida State Univ. College of Eng., Tallahassee, FL)

The purpose of this study was to examine the acoustic and electroglottographic features in the characterization of passaggio in female singers. Three groups of female singers were instructed to sing the notes of the scale for one octave using an “ah” vowel. When singing this octave they sang through a register shift, which is called “passaggio.” Their singing voices were recorded in a two-channel dataset. The first channel captured the acoustic signal using a microphone, and the second channel captured the electroglottographic (EGG) signal using an EGG instrument. This study used VoiceSauce (Shue, Keating, and Vicenik, 2009) analysis software to analyze the features of the two-channel dataset that contribute to the characterization of the passaggio in female singers. Glottal measurements can give more robust information on precise glottal opening and closing moments using the derivative of the EGG signal (DEGG). The results of this investigation also provide an analytical framework for calculating the DEGG in female singers.

4:00

1pMU10. A study of the type and characteristics of relaxing music for college students. Wei-Chun Wang (National Taiwan Univ. of Sci. and Technol., No. 43, Sec. 4, Keelung Rd., Taipei 106, Taiwan, vgnwang@hotmail.com)

It is believed that music has the power to soften emotions and alleviate pains. The essence of the power has being encoded by researchers. This study was aimed to explore the effects of music preference and stress-associated responses of college students when they listen to music. The objectives of this study were (1) to survey the music types of relaxing music for college students, and the difference in gender and study majors; (2) to investigate the effects of musical preference, music expertise, and awareness of musical content on their perceptivity of relaxation; and (3) to analyze the relativities of musical emotions with musical characteristics, such as tempo, mode, and dynamic range. Participants were asked to listen to selected music pieces, and to rate their three-dimensional emotional responses, pleasant-unpleasant, calm-arousal, and relaxing-stress, on five-point Likert scales. Data collection of music compositions and personal music taste was acquired using surveys. The findings are expected to (1) understand college students’ listening habit, (2) collect repertoire suitable for university students in terms of stress releasing, and (3) offer advices for music appreciation teaching, psychological consultation personnel, and clinical therapist.

Session 1pNS

Noise, Architectural Acoustics, and ASA Committee on Standards: Soundscapes: Decisions on Measurement Procedures

Brigitte Schulte-Fortkamp, Cochair
TU Berlin, Einsteinufer 25 TA 7, Berlin 10587, Germany

Klaus Genuit, Cochair
HEAD Acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany

Chair's Introduction—1:00

Invited Papers

1:05

1pNS1. Sounding Brighton: Update on soundscape planning with a user centric approach. Lisa Lavia (Noise Abatement Society, 8 Nizells Ave., Hove BN3 1PL, United Kingdom, lisa.lavia@noise-abatement.org)

The potential of soundscape planning has been widely illustrated in recent years. Sounding Brighton is a collaborative initiative pioneered by the Noise Abatement Society and Brighton and Hove City Council in 2010 with the support of the former COST Action TD0804. The project continues and is exploring the positive effects soundscapes can have on health, wellbeing, and quality of life. It recently undertook a city-wide soundscape survey and interviews leading to "West Street Story," a night-noise intervention pilot, to gauge whether ambient soundscapes might act as an antidote to the Saturday night drinking culture seen on the city's most dangerous street, and the follow on project: "West Street Tunnel" investigating the same approach in a disused pedestrian subway. The work has also gained inclusion in the Masterplan for the redevelopment of the city center, leading to its willingness to participate in and its acceptance into the European Union funded FP7 SONORUS project looking at holistic ways to include urban sound planning into city planning. This paper will provide an update of the project and its results so far.

1:25

1pNS2. Applicability of measurement procedures in soundscape context—Experiences and recommendations. Klaus Genuit and Fiebig André (HEAD Acoustics GmbH, Ebertstr. 30a, Herzogenrath 52134, Germany, klaus.genuit@head-acoustics.de)

In the context of the ISO/TC 43/SC 1/WG 54, different aspects of soundscape will be subject to standardization. Besides a standardized soundscape definition, understanding and terminology, minimum reporting requirements for measuring soundscapes including measurement uncertainties are currently discussed and will be subject to standardization later. All in all, a wide range of measurement procedures is applied for measuring, describing, documenting, and analyzing soundscapes. However, several aspects of and conditions for measurements are still unclear, which limits the comparability and compatibility of soundscape investigations. It is evident that a common basis of measurement procedures is needed to bring forward current standardization efforts. Consequently, it is very important to share experiences and knowledge about measurement procedures and their general applicability in soundscape context. A thorough discussion about the data quality achieved by certain measurement procedures is inevitable as well. Observations and experiences made in different soundscape studies regarding different measurement procedures will be presented and discussed with respect to their significance and applicability.

1:45

1pNS3. The need for a soundscape taxonomy. Brigitte Schulte-Fortkamp (Inst. of Fluid Mech. and Eng. Acoust., TU Berlin, Einsteinufer 25, Berlin 10587, Germany, b.schulte-fortkamp@tu-berlin.de)

It is for some time now that the standardization process in Soundscape in the ISO/TC 43/WG 54 12913 has started. Meanwhile, it has reached the level of decisions regarding evaluation approaches. Hence, as perception is the foreground of any assessment procedures here, on the one hand interviews do play a major role, but also procedures as sound walks. On the other hand, a time to evaluation related psychoacoustic measurement is needed to bring relevant datasets into triangulation. This paper will present the current state of the art in Soundscape and will discuss further input that is needed in this field.

2:05

1pNS4. Comparison of momentary and retrospective soundscape evaluations. Jochen Steffens (School of Information Studies & CIRMMT, McGill Univ., Josef-Gockeln-Strasse 9, Duesseldorf 40476, Germany, jochen.steffens@fh-duesseldorf.de), Johannes Petrenko (Inst. of Sound and Vib. Eng., Univ. of Appl. Sci., Duesseldorf, Germany), and Catherine Guastavino (School of Information Studies & CIRMMT, McGill Univ., Montreal, QC, Canada)

The Peak and End rule describes the effect that retrospective evaluations of temporal events significantly depend on the most extreme affect (peak) experienced during an episode and on the affect at the ending. Other features, like the duration of an event, seem to be widely negligible. We are testing this hypothesis in the context of soundscape evaluation in a series of listening tests conducted in MontrÕal, Canada, and Duesseldorf, Germany. The soundscapes consisted of recordings of different locations (e.g., railway station, park) and were edited so that there was one presumed emotional "peak moment." The task of the test group was to indicate momentary judgments by continuously adjusting a slider on a computer interface over the course of the stimulus presentation. Additionally, the participants had to make an overall retrospective rating of the soundscapes after listening to them. To investigate attention effects in the course of the test task a second group was asked to only judge the sounds retrospectively. Preliminary results indicate that the Peak and End rule in combination with the averaged momentary evaluations well predict the retrospective judgments. Within this contribution the results of the experiment will be presented and implications for soundscape design will be discussed.

2:25

1pNS5. Case studies of historic soundscapes with cultural importance. David Lubman (dlAcoustics, 14301 Middletown Ln., Westminster, CA 92683, dlubman@dlacoustics.com)

Case studies discussed include Bow Bells, in which the sounds of ringing bells of the East London church of St. Mary Le Bow became a soundmark that defined the "Cockney" ethnicity. (A "Cockney" is a person born within the sound of Bow Bells.) With London's growth, more churches were built. The combined sounds of their bells became the community soundscape for East London. It inspired beloved children's poetry (Oranges and Lemons ? Says the great Bell of Bow). That children's poem was learned by generations of English children, and the poem became the sound symbol for all of England. This became very important in WWII when Nazi arial bombing destroyed Bow Bells creating a serious morale problem for England. To reassure their countrymen that the "home fires" were still burning, BBC used a 1926 sound recording of Bow Bells as an interval signal in their worldwide radio broadcasts. On a smaller scale, accidental whispering arches can give buildings "a sense of place." A 6th C monastery at Clonmacnoise, Ireland, has a large and elaborately carved door installed that became a storied whispering archway. The storied accidental whispering archways at Grand Central Terminal and Pennsylvania's West Chester University are also described.

Contributed Papers

2:45

1pNS6. Cross-modal soundscape mapping: Integrating ambisonic field recordings with high dynamic range spherical panoramic photography to produce interactive maps. J. Parkman Carter (Architectural Acoust., Rensselaer Polytechnic Inst., 32204 Waters View Circle, Cohoes, NY 12047, cartej8@rpi.edu) and Jonas Braasch (Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

We cannot "measure" the soundscape any more than we can "measure" the ocean, the city, or the wilderness. Being comprised of myriad complex elements, conditions, and relationships between sound sources and sound perceivers, the soundscape—and any sufficient description of it—must account for several different, but significantly interrelated, dimensions: physical, spatial, temporal, perceptual, cultural, and historical. How, then, are we to meaningfully document the soundscape? If we are to begin to understand the soundscape's impact on us—and our impact upon it—we need new methods to capture and represent the multisensory extents of a soundscape without reverting to one-dimensional quantitative abstractions. This project proposes an interdisciplinary method to record a soundscape's multisensory attributes by combining aural and visual information in a structured way which links the directionality of view and sound arrival. This method integrates multi-directional Ambisonic audio recordings with high dynamic range (HDR) spherical panoramic photography in the form of interactive maps and virtual tours. Case studies using the cross-modal soundscape mapping method will be presented.

3:00

1pNS7. The sound pressure level observing transponder: A satellite-enabled sensor package for near real time monitoring. Peter Marchetto, Christopher W. Clark (BioAcoust. Res. Program, Cornell Lab. of Ornithology, Cornell Univ., 206 Langmuir Lab., 95 Brown Rd., #1012, Ithaca, NY 14850, pmm223@cornell.edu), and Daniel Aneshansley (Biological and Environ. Eng., Cornell Univ., Ithaca, NY)

Many *in situ* sensing applications for bioacoustic ecology have suffered from a lack of means to communicate information in near real time. The monitoring of incident noise on an individual animal and its behavioral response to it were the focus of this project. The sensor platform described herein may be used to create acoustic field maps of a habitat in near-realtime.

3:15–3:45 Panel Discussion

Session 1pPP

Psychological and Physiological Acoustics: From Protection to Perception

Charlotte M. Reed, Chair
Massachusetts Inst. of Technol., 77 Massachusetts Ave., Rm. 36-751, Cambridge, MA 02139

Chair's Introduction—1:00

Contributed Papers

1:05

1pPP1. Measurement of hearing-protector attenuation using auditory steady state responses. Olivier Valentin (Dept. of Mech. Eng., ècole de technologie supérieure, 1100 Rue Notre-Dame Ouest, Montréal, QC H3C 1K3, Canada, m.olivier.valentin@gmail.com), Michael Sasha John (Rotman Res. Institute/Inst. of BioMater. and Biomedical Eng., Univ. of Toronto, Toronto, ON, Canada), and Frédéric Laville (Dept. of Mech. Eng., ècole de technologie supérieure, Montréal, QC, Canada)

There is a need to assess hearing protection device (HPD) attenuation to ensure that individuals receive adequate protection from noise. Present methods of attenuation measurement have limitations. Objective measurements such as field microphone in real ear (F-MIRE) do not assess bone conducted sound. Psychophysical measurements such as real ear attenuation at threshold (REAT) are biased due to the low frequency masking effects from test subjects' physiological noise, and the variability of measurements based on subjective response. We explored using auditory steady state responses (ASSR) as a technique that may overcome these limitations. ASSRs were recorded in ten normal hearing adults, using both "normal" and "occluded" conditions. Stimuli included both narrow band noises and pure tones (500 and 1000 Hz), amplitude modulated at 40 Hz. Stimuli were presented through either loudspeakers or headphones, at 45, 55, and 65 dB SPL. "Physiological attenuation" was calculated as the difference between ASSR values (both amplitude and phase) for normal and occluded conditions. Physiological attenuation estimates were compared to in-ear (objective) and psychophysical (subjective) measurements. Grand mean ASSR data complied well with in-ear and subjective measurements. Further work is needed to provide accurate assessment of ASSR-based physiological attenuation for individual subjects.

1:20

1pPP2. A public science experiment on the link (if any) between hearing loss and a lifetime of loud music exposure. Michael Akeroyd, William 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), Robert Mackinnon, Heather Fortnum (NIHR Nottingham Hearing Biomedical Res. Unit, Univ. of Nottingham, Nottingham, United Kingdom), and David R. Moore (Commun. Sci. Res. Ctr., Cincinnati Childrens Hospital, Cincinnati, OH)

It is well known that a lifetime of exposure to loud industrial noise can lead to substantial amounts of hearing damage. The effect of a lifetime of loud music exposure is far less certain, however. To assess if there is such a link, we have launched a public science experiment that combines (a) a questionnaire on the amount of listening to loud music over one's life as well as standard questions on their hearing, and (b) a triple-digit speech-in-noise test specifically crafted for high-frequency hearing losses. The experiment has a dedicated website and takes advantage of the 2013 Centenary of the Medical Research Council to act as a "pull" for the public to participate. Over the 100 years, the MRC has been funding research, there has been revolution after revolution in the technology for music reproduction for the home or stage and in the volumes they can reach. The talk will describe the design of the experiment, the results so far, and our experience-based suggestions for running future web-based, public-science experiments. [Work supported by the Medical Research Foundation, the Medical Research Council, and the Chief Scientist Office, Scotland.]

1:35

1pPP3. A novel mode of off-frequency hearing as a result of defective outer hair cells hair bundles unveiled by Nherf1-/- mice. Aziz El-Amraoui, Kazusaku Kamiya, Vincent Michel (Neurosience, Genetic and Physiol. of Audition, Institut Pasteur, 25 rue du Dr Roux, Paris 75015, France, aziz.el-amraoui@pasteur.fr), Fabrice Giraudet (Faculté de Médecine, Université d Auvergne, Clermont-Ferrand, France), Maria-Magdalena Gerogescu (Neuro-Oncology, Texas M.D. Anderson Cancer Ctr., Houston, TX), Paul Avan (Faculté de Médecine, Université d Auvergne, Clermont-Ferrand, France), and Christine Petit (Neurosience, Genetic and Physiol. of Audition, Institut Pasteur, Paris, France)

Nherf1, a PDZ-domain-containing protein, was identified in the hair bundle in differentiating outer hair cells (OHCs). Nherf1-/- mice showed apparently mild hearing-threshold elevations at mid/high sound frequencies, associated to OHC hair-bundle shape anomalies, prominent in the basal cochlea. This mild impact on hearing sensitivity was discordant with the finding of almost non-responding OHCs in the basal cochlea as assessed by distortion-product otoacoustic emissions and cochlear microphonic potentials. Unlike normal mice, responses of Nherf1-/- mice to high-frequency test tones were not masked by tones of neighboring frequencies. Efficient masker tones displayed unusual characteristics: maximal efficiency at lower frequencies (up to two octaves lower than the test tone), and at low levels (up to 25 dB below test-tone level). This, and the relative growth of the masker and test tones, suggests that mid-high frequency tones of moderate intensity are detected off-frequency, in the functionally unaffected apical cochlear region. Our results establish that Nherf1 is critical for hair bundle morphogenesis and reveal a novel mode of off-frequency detection, probably involving the persistent contact between OHCs and the tectorial membrane. These findings suggest how to circumvent major pitfalls in hearing assessment of some patients, by avoiding misleading interpretations of hearing thresholds.

1:50

1pPP4. Evidence against power amplification in the cochlea. Marcel van der Heijden and Corstiaen Versteegh (Neurosci., Erasmus MC, P.O.Box 2040, Rotterdam 3000 CA, Netherlands, m.vanderheyden@erasmusmc.nl)

Sound-induced traveling waves in the mammalian inner ear peak at a frequency-dependent location. Some form of motility is widely believed to boost this peaking by injecting extra power into the wave. We determined the power carried by the wave from two-point recordings of basilar membrane motion in sensitive cochleae. Up to moderate intensities, the peak wave power was slightly less than the acoustic power entering the middle ear. At higher intensities, an increasingly smaller fraction of the acoustic

power reached the peak region. Thus, cochlear dynamic compression stems from variable dissipation rather than saturating amplification. Additional measurements revealed that the peaking of the wave envelope is realized by focusing the acoustic power rather than amplifying it.

2:05

1pPP5. Effects of self-generated noise on estimates of detection threshold in quiet in school-age children and adults. Emily Buss, Heather L. Porter (Otolaryngol.—Head and Neck Surgery, UNC Chapel Hill, 170 Mannng Dr., G190 Physicians Office Bldg., CB# 7070, Chapel Hill, NC 27599, ebuss@med.unc.edu), Lori J. Leibold (Allied Health Sci., UNC Chapel Hill, Chapel Hill, NC), John H. Grose, and Joseph W. Hall (Otolaryngol.—Head and Neck Surgery, UNC Chapel Hill, Chapel Hill, NC)

Detection in quiet develops earlier in childhood for high than low frequencies. The present study tested the hypothesis that self-generated noise could play a role in this finding. When adults listen for sounds near threshold, they tend to engage in behaviors that reduce physiologic noise (e.g., quiet breathing), which is predominantly low frequency. Children may not suppress self-generated noise to the same extent as adults. This possibility was evaluated by measuring sound levels in the ear canal simultaneous with adaptive threshold estimation for 250-, 1000-, and 4000-Hz pure tones. Stimuli were delivered and recordings were made using a single foam insert. Listeners were children (4.3–16.0 yr) or adults. Consistent with previous data, the effect of child age was robust at 250 Hz, whereas thresholds of even the youngest listeners were nearly adult-like at 4000 Hz. The spectral shape of self-generated noise was generally similar across listener age groups, although the magnitude was higher in younger listeners. Trial-by-trial data were evaluated to assess the relationship between noise and the accuracy of listener responses: there was an association for younger listeners. These results provide preliminary evidence that self-generated noise may play a role in the prolonged development of low-frequency detection in quiet.

2:20

1pPP6. Adaptation reveals automatic pitch-shift detectors. Samuel R. Mathias, Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215, smathias@bu.edu), Christophe Micheyl (Starkey Hearing Res. Ctr., Berkeley, CA), and Andrew J. Oxenham (Dept. of Psych., Univ. of Minnesota, Minneapolis, MN)

Previous work suggests that the auditory system contains automatic frequency- or pitch-shift detectors (PSDs). These hypothetical units may serve to bind successive sounds together, and appear to be tuned to optimally detect frequency shifts of about 1 semitone. The present work describes several experiments that provide more evidence for PSDs. In all experiments, listeners judged the direction (up or down) of small relevant frequency shifts whilst ignoring prior irrelevant shifts that were usually much greater in magnitude. In almost all conditions and experiments, there was a nonmonotonic relationship between sensitivity to the relevant shift (d') and the magnitude of the irrelevant shift (Δ). d' declined as a function of increasing Δ up to about 1 semitone, and further increases in Δ increased d'. The "dip" in d' could not be explained by selective attention, frequency-specific adaptation, or energetic masking. PSDs provide a convenient explanation for these results if one assumes that (a) listeners discriminate small frequency shifts using PSDs, (b) PSDs are automatically activated by irrelevant frequency shifts, causing them to adapt, and (c) maximal adaptation of PSDs occurs at around 1 semitone.

2:35

1pPP7. Effects of presentation method and duration on alarm detection threshold in the presence of loud pink noise. Buddhika Karunarathne, Richard So (Industrial Eng. & Logistics Management, Hong Kong Univ. of Sci. & Technol., Clear Water Bay, Kowloon, Hong Kong, Kowloon 00000, Hong Kong, jbpdk@ust.hk), and Anna Kam (Dept. of Otorhinolaryngology, Head & Neck Surgery, Chinese Univ. of Hong Kong, Hong Kong, Hong Kong)

Detection of pure tone signals in the presence of noise has been thoroughly studied. Most of these studies have used monaural presentation of audio stimuli. Also, studies testing alarm detection in the presence of noise are limited. In 2013, Karunarathne *et al*., conducted a study and found out that human listeners were able to detect an alarm in negative signal-to-noise ratios (SNRs), as low as −-24 dB. This study aims to investigate the effects of presentation method and duration of the alarms on detection threshold. Eight conditions varied by presentation method (monaural vs spatial) and alarm duration were tested. Sixteen human subjects with normal hearing were given the task of identifying which one of two sound intervals contained an alarm along with 80dBA pink noise. Thresholds were estimated as the 79.4% points on the psychometric functions, using adaptive 2-Interval Forced Choice (2IFC) procedure with a 3-down 1-up rule. Results indicated that detection thresholds were statistically significantly lower in spatial condition compared to monaural. The effect of alarm duration was not significant in both spatial and monaural conditions. Thresholds lower than −-30 dB SNR were observed in the spatial condition, which agreed with the findings of Karunarathne *et al*. and further extended threshold boundaries.

2:50

1pPP8. Predictions of the magnitude of tonal content on the basis of partial loudness. Jesko L. Verhey, Jan Hots (Dept. of Experimental Audiol., Otto von Guericke Univ., Leipziger Str. 44, Magdeburg 39120, Germany, jesko.verhey@med.ovgu.de), and Matthias Vormann (Hörzentrum Oldenburg GmbH, Oldenburg, Germany)

Environmental sounds containing clearly audible tonal components are considered to be more annoying than sounds without these components. Several standards include sections dedicated to the assessment of tonal components in sound. These standards have in common that they estimate the magnitude of the tonal components (in the following referred to as tonalness) as the level above the noise background. Recent studies indicate that partial loudness of the tonal component determines the tonalness. The present study tests this hypothesis by comparing experimental data on tonalness of sounds with multiple tonal components with predictions of a loudness model. It is shown that partial loudness is a better predictor of the perception of tonal portions of a sound than the intensity. This approach may be useful for future standardization of the perception of clearly audible tonal components in noise.

3:05–3:20 Break

3:20

1pPP9. The unknown effects of amplitude envelope: A survey of *Hearing Research*. Jessica Gillard (Psych., Neurosci. & Behaviour, McMaster Inst. for Music and the Mind, 1280 Main St. West, Hamilton, ON L8S 4M2, Canada, gillarj@mcmaster.ca) and Michael Schutz (School of the Arts, McMaster Inst. for Music and the Mind, Hamilton, ON, Canada)

In auditory research, the use of amplitude-steady tones with abrupt onsets and offsets is quite common. While these types of "flat" tones offer a great deal of control, they are not representative of the types of sounds we hear outside the laboratory. In everyday listening we are much more likely to encounter "percussive" (i.e., exponentially decaying) sounds, with offsets conveying detailed information such as the materials and force used to produce the sounds—information that is absent in abruptly ending flat tones. Given that differences in perception have been reported when using different amplitude envelopes (Grassi and Pavan, 2012; Neuhoff, 1998; Schutz, 2009), we became interested in determining the prevalence of flat and percussive tones in auditory research publications. Here, we surveyed the journal Hearing Research and classified the temporal structure of sounds used into five categories: flat, percussive, click train, other, and undefined. We found 42.5% of sounds were flat (approximately 13% were click trains, 4% other, and 40% undefined). This finding is consistent with our previous surveys of *Music Perception* and *Attention, Perception, and Psychophysics*, suggesting that flat tones dominate auditory research, and the perceptual effects of more naturalistic sounds are relatively unknown and ripe for future exploration.

3:35

1pPP10. Could binaural listening help segregate competing sound sources? Marion David (Laboratoire Génie Civil et Bâtiment, ENTPE, rue Maurice Audin, Vaulx-en-Velin 69518, France, marion.david@entpe.fr), Nicolas Grimault (Ctr. de Recherche en NeuroSci. de Lyon, Université de Lyon, Lyon, France), and Mathieu Lavandier (Laboratoire Génie Civil et Bâtiment, ENTPE, Vaulx-en-Velin, France)

When a sound is alternatively played from two positions, the resulting sequence at one ear is an alternate of two spectrally different sounds, due to head coloration. A previous study showed that these monaural spectral differences can induce segregation. Here, binaural cues are introduced to investigate whether they could strengthen segregation. A rhythmic discrimination task evaluated obligatory streaming with speech-shaped noises. In a first experiment, head-related transfer-functions were modified to introduce independently the interaural time and level differences (ITD and ILD). The results suggested that both ITD and ILD favored segregation. Moreover, the perceptive organization was rather based on the monaural variations in spectrum and intensity at one ear rather than on the spectral interaural differences. Since the binaural cues allow the auditory system to lateralize sounds, a second experiment was intended to determine to which extent the influence of ITD was due to the interaural difference and/or to the resulting associated perceived position. Temporal delays were introduced to simulate different ITDs. The perceived position was modified by manipulating these ITDs independently across frequency. The results, combined with a subjective test of lateralization, showed that both ITD and perceived position influence stream segregation.

3:50

1pPP11. Feedback loops in engineering models of binaural listening. Jens P. Blauert, Dorothea Kolossa (Inst. of Commun. Acoust., Ruhr-Universitaet Bochum, Bochum 44780, Germany, jens.blauert@rub.de), and Patrick Danés (LAAS-CNRS, Univ. Toulouse III Paul Sabatier, Toulouse, France)

Hearing models for tasks like auditory scene analysis or sound-quality judgments can run into severe problems when acting in a purely bottom-up, that is, signal driven manner, as they may have to follow up on all possible output options until a final decision has been taken. This may lead to a combinatorial explosion. A way out is the inclusion of top-down, that is, hypothesis-driven processes. In top-down processing, the number of states to be evaluated can be reduced substantially, when the system knows what to look for and thereby focuses attention on states which make sense in a given specific situation. To implement adequate top-down processes, various feedback loops will be included in our models, some more specific, others more general. The general ones originate from the concept that the listener model ("artificial listener") actively explores acoustic scenes and thereby develops its aural world in an autonomous way. Following this notion, it is attempted to model listeners according to the autonomous-agents paradigm, where agents actively learn and listen. [Work performed in the context of the EU project TWO!EARS,<www.twoears.eu>.]

4:05

1pPP12. Quantifying the better ear advantage in the presence of interfering speech. Esther Schoenmaker and Steven van de Par (Acoust. Group, Univ. of Oldenburg, Carl von Ossietzkystrasse 9-11, Oldenburg D-26129, Germany, esther.schoenmaker@uni-oldenburg.de)

In cocktail party listening with spatially separated speech sources, better ear listening is known to make a major contribution to speech intelligibility. The better ear is generally defined as the ear that receives the highest signal-to-noise ratio (SNR). Usually, this SNR is calculated based on the total length of the signal. However, this seems inappropriate when speech signals are involved since these are highly modulated both in the time and frequency domain. On a perceptual level, modulated maskers give rise to a higher target speech intelligibility than their unmodulated counterparts through the presence of glimpses. A simple measure to quantify the better ear advantage while taking these spectrotemporal fluctuations into account is introduced. In a headphone experiment, three simultaneous sequences of vowel-consonant-vowel utterances were presented at a fixed target-to-masker ratio. The stimuli were rendered with head-related transfer functions and contrasted against stimuli that did not contain any interaural level differences (ILDs) and, as a consequence, allowed no better ear listening. Using the proposed metric, we are able to explain differences in intelligibility for these speech-in-speech mixtures that would remain unexplained by the conventional SNR both for stimuli with and without ILDs.

4:20

1pPP13. Relative importance of individual spectral features for intraconic localization. Griffin D. Romigh (711th Human Performance Wing, Air Force Res. Labs, 4064 chalfonte, beavercreek, OH 45440, griffin.romigh@wpafb.af.mil), Brian D. Simpson (711th Human Performance Wing, Air Force Res. Labs, Dayton, OH), Eric R. Thompson (Ball Aerosp. & Technologies Corp., Dayton, OH), and Nandini Iyer (711th Human Performance Wing, Air Force Res. Labs, Dayton, OH)

Most researchers agree that physical features present in the monaural spectra act as the primary set of cues for sound source localization within a cone-of-confusion. Less consensus has been reached as to whether these localization judgments are based on the presence of simple spectral features, or whether a more broad spectrum pattern matching occurs in which information across many bands are utilized. The present work first describes a head-related transfer function decomposition technique by which spectral cues are separated into components utilized for lateral localization judgments (when combined binaurally) and those used for localization judgments within a cone-of-confusion. This decomposition allows us to modify individual spectral features of an arbitrary virtual stimulus while maintaining its naturalness and perceived lateral location, criteria that were not adhered to in previous studies. Using this technique, a set of virtual localization studies was conducted in which individual spectral features were removed to examine their relative importance to intraconic localization judgments. Results indicate that while both spectral peaks and notches contribute to localization judgments, spectral information appears to be integrated across multiple frequency bands.

4:35

1pPP14. The role of modulation processing in binaural masking-patterns. Bjoern Luebken (Dept. of Experimental Audiol., Otto von Guericke Univ., Leipziger Str. 44, Magdeburg 39120, Germany, bjoern.luebken@med.ovgu.de), Steven van de Par (Acoust. Group, Carl von Ossietzky Univ., Oldenburg, Germany), and Jesko L. Verhey (Dept. of Experimental Audiol., Otto von Guericke Univ., Magdeburg, Germany)

Binaural masking pattern experiments indicate a continuous decrease in the binaural masking-level difference (BMLD) with increasing spectral distance of a tonal signal to a narrowband noise masker. Previous studies suggested that this decrease in BMLD is due to additional modulation cues in monaural off-frequency masking conditions. An own masking pattern experiment with an additional interferer to mask modulation cues supported this hypothesis. The interferer is positioned spectrally below/above the masker for the signal above/below the masker with a spectral distance equal to the distance between masker and signal. This interference tone has a large impact on the thresholds without a binaural cue, as expected. Such an increase in the diotic thresholds is predicted on the basis of a modulation-filterbank model, but only if an across-channel modulation processing is assumed. The interferer also increases the dichotic thresholds, indicating an influence of modulations processing also in conditions where binaural cues are present. Assuming that modulation cues are masked by the interference tone and thus the detection is based on energy cues, the binaural data indicate effectively wider binaural filter, as previously suggested on the basis of notched-noise experiments.

4:50

1pPP15. Hearing better with interaural time differences and bilateral cochlear implants. Zachary M. Smith (Res. & Technol. Labs, Cochlear Ltd., 13059 E Peakview Ave., Centennial, CO 80111, zsmith@cochlear.com), Alan Kan, Heath G. Jones, Melanie Buhr-Lawler, Shelly P. Godar, and Ruth Y. Litovsky (Univ. of Wisconsin Waisman Ctr., Madison, WI)

While bilateral cochlear implant (BiCI) recipients generally receive significant benefits from the addition of a second ear, evidence suggests that much of the benefit is attributed to monaural effects or to the availability of interaural level differences. The benefits are less than those in normal-hearing listeners, however, in part because the sound localization and speech unmasking that are measured may show greater benefits if interaural time difference (ITD) cues were also available. To date, there is a paucity of evidence that ITDs can be captured and saliently delivered by cochlear implant processors. In this study, we used a research processing strategy that explicitly codes ITD cues. We measured BiCI listeners' ITD sensitivity to broadband speech material and ITD-based unmasking in a multi-talker listening scenario by directly presenting sounds through the accessory inputs of their sound processors. Performance was compared to that with the commercial ACE processing strategy. Initial results show that some subjects with good ITD sensitivity can also take advantage of ITD to better understand a target talker in the presence of a masking talker at low target-to-masker ratios. This suggests that improving ITD perception in BiCIs may lead to better hearing outcomes in real-world listening situations.

5:05

1pPP16. Resolution and integration within auditory temporal windows. Xiangbin Teng, Xing Tian, and David Poeppel (Psych., NYU, 6 Washington Pl., New York, NY 10003, david.poeppel@nyu.edu)

Temporal integration in auditory and speech perception is investigated in a variety of experimental contexts, using different types of signals. How the auditory system manages the tension between resolution, on the one hand, and integration, on the other—and in particular how the system integrates acoustic information over time to form a unitary percept—remains unclear. Using non-speech signals with temporal structure at different scales (30 ms, 200+ ms), we tested in four psychophysical experiments how "local" (shorter-scale) and "global" (longer-scale) auditory information is resolved and integrated. We provide evidence (supported by independent electrophysiological data) that the auditory system extracts temporally detailed acoustic information using small temporal windows, and then integrates that information over approximately 200 ms. Importantly, the fine-detailed information is not fully accessible as sound duration increases to over ~100 ms. Further, we show that representation of acoustic information requires ~150–200 ms, and that the representation of fine-detailed information can compromise temporal integration. The findings demonstrate the time scales over which the integration and resolution of auditory information cooperate and conflict, and thus shed light on the mechanisms of temporal integration.

MONDAY AFTERNOON, 5 MAY 2014 553 A/B, 1:25 P.M. TO 4:25 P.M.

Session 1pSA

Structural Acoustics and Vibration and Underwater Acoustics: Undersea Vehicle Noise

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

Nickolas Vlahopoulos, Cochair
Univ. of Michigan, 2600 Draper Dr., Ann Arbor, MI 48109

Chair's Introduction—1:25

Invited Papers

1:30

1pSA1. Acoustic signature of underwater vehicles. Joe M. Cuschieri (Lockheed Martin MST, 100 East 17th St., Riviera Beach, FL 33404, joe@cuschieri.us)

Underwater vehicles are becoming more commonly used for a number of applications ranging from commercial to defense. While the acoustic signature of underwater vehicles is understood to be important, there are presently more pressing issues which are fundamental to the success of using underwater vehicles beyond the lab or controlled environment. Such issues incorporate power sources, navigation accuracy, reliable control, etc. However, while acoustic signature may not be in the forefront, need for low acoustic signature and low self-noise is important as the noise form the underwater vehicle can impact the operation of the acoustic sensors. In this paper, components that influence the radiated noise from underwater vehicles are identified and discussed, especially as these relate to their impact on the overall acoustic radiation. Approaches to estimate the underwater radiated noise based on other data when in water data is not available are discussed.

1:50

1pSA2. Decreasing the radiated acoustic and vibration noise of both prop-driven and buoyancy-driven autonomous underwater vehicles. Richard Zimmerman, Gerald L. D'Spain (Marine Physical Lab, Scripps Inst. of Oceanogr., 291 Rosecrans St., San Diego, CA 92106, gdspain@ucsd.edu), Peter Brodsky (Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Mark Stevenson (SPAWAR SSC Pacific, San Diego, CA), Mark Zumberge, and John Orcutt (Marine Physical Lab, Scripps Inst. of Oceanogr., San Diego, CA)

Our previously published results from decreasing the radiated acoustic and vibration noise of a mid-size, prop-driven autonomous underwater vehicle (AUV) show self noise levels recorded at sea by an AUV-mounted hydrophone array that are at, or below, typical background ocean noise levels across the frequency band above 200 Hz. The remaining noise below 200 Hz is primarily vibration induced. The modifications required to achieve this 20–50 dB reduction in propulsion and steering system noise levels will be reviewed in this talk. In addition, at-sea measurements of the acoustic noise radiated by the large (30 L) buoyancy engine on a 20-ft wing span flying wing autonomous underwater glider are presented. Whereas a prop-driven system operates continuously, the buoyancy-driven propulsion system has a very low duty cycle of a few percent; it is on only for about 3 min during each dive cycle. Onboard self noise from the glider's internal fluid-based roll control system far exceeds that from an aileron-based system. However, the former system provides control authority at or near neutral buoyancy. [Work supported by the Office of Naval Research and BP.]

2:10

1pSA3. Aspect-dependent acoustic characterization of an underway autonomous underwater vehicle. John Gebbie, Martin Siderius (Northwest Electromagnetics and Acoust. Res. Lab., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, jgebbie@ece.pdx.edu), and John S. Allen (Dept. of Mech. Eng., Univ. of Hawai i-Manoa, Honolulu, HI)

Acoustic emissions emitted by an underway REMUS-100 autonomous underwater vehicle (AUVs) are analyzed with array beamforming. Characterizing emissions of underway AUVs is challenging due to a variety of factors such as low source levels of some vehicle types, continually varying propagation conditions, and inherent uncertainties in vehicle location. While aspect-dependent spectral and source level emissions are available for a wide range of surface craft types, few articles have analyzed these for underway AUVs. Array beamforming is a known method of increasing gain of a weak signal in the presence of interference and noise, and propagation modeling tools can provide estimates of transmission loss between a source and receiver. These techniques are used to measure the aspect-dependent source level and spectral content from the propulsion system of a REMUS-100 AUV deployed near a fixed array near Honolulu Harbor, Hawaii, by fusing measured acoustic data with multi-sensor navigational records recorded on the vehicle. As AUVs become more widely used, characterizing their acoustic radiation will help predict how these platforms interact with and impact the environments in which they are deployed.

2:30

1pSA4. Concurrent control and plant identification to support overall noise reduction of autonomous underwater vehicles. Jason D. Holmes, Alison B. Laferriere, and Christopher G. Park (Sensor Systems, Raytheon BBN Technologies, 10 Moulton St., Cambridge, MA 02375, jholmes@bbn.com)

Commercial, off-the-shelf (COTS) autonomous underwater vehicles (AUVs) are not usually designed to be low-noise in the 100 Hz–10 kHz band. Options to mitigate the noise include mechanical re-design, adaptive filtering of the noise on any acoustic sensor using the vehicle as a platform, or active noise control (ANC). The first of these options eliminates the benefits of using a COTS platform. The second option can be attractive but is most effective for propulsor noise (and not other sources like the depth sounder) and does not mitigate the issue if multiple vehicles are used in coordination. An on-board projector can enable ANC of the vehicle noise. For small, nearly acoustically compact vehicles, the directionality of propulsor noise is not very complex, suggesting global control using a limited set of the acoustic sensors as a control metric. This paper explores methods for performing concurrent control of narrow-band vehicle noise while performing the low-noise, broad-band plant identification necessary to support the control. In addition to the reductions in self noise (on a single vehicle and multiple vehicles), the ability to use the information in the plant estimate to perform "quiet" depth sounding is explored.

2:50

1pSA5. Modeling the acoustic color of undersea structures. David Burnett (Naval Surface Warfare Ctr., Code CD10, 110 Vernon Ave., Panama City, FL 32407, david.s.burnett@navy.mil)

The Naval Surface Warfare Center Panama City Division has developed a 3-D finite-element computer simulation system, PC-ACOLOR, for modeling the "acoustic color" (target strength as a function of frequency and aspect angle) of single and multiple realistic elastic structures that are near to or straddling the water/sediment interface at the bottom of the ocean. Target strength is a measure of the intensity of scattered fields, but radiated noise can also be modeled, yielding sound pressure level as a function of frequency and aspect angle. PC-ACOLOR employs 3-D continuum mechanics throughout the entire computational domain; no engineering approximations are used anywhere. The talk will give an overview of PC-ACOLOR: important structural acoustic concepts, e.g., the relationship between 3-D modeling and evanescent wave solutions to the Helmholtz pde; modeling techniques; verification and validation; and examples of different types of vehicles and other structures that have been modeled.

3:10–3:30 Break

3:30

1pSA6. A computational approach to flow noise. Donald Cox, Daniel Perez, and Andrew N. Guarendi (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, donald.l.cox@navy.mil)

The ability to calculate the noise internal to a structure due to external flow is a necessity for the optimal design of a low noise structure. Whether the structure is an automobile, an airplane or an acoustical array, the goal is the same: to in some way minimize the acoustic pressure/particle velocity resulting from flow excitation at a design location. The excitation is usually that due to wall pressure fluctuations resulting from turbulent boundary layer (TBL). There is a long history of modeling plate excitation due to TBL loads. In most cases, the existing work makes use of statistical, empirically based models for the TBL excitation. This work focuses on combining the capabilities of computational fluid dynamics with computational structural acoustics to enable the calculation of flow noise primarily for undersea vehicles. The work is limited to the non-coupled problem, where the flow calculations are made over a non-deforming boundary with the goal of calculating wall pressure fluctuations and using them as loads on a finite element structural acoustics model. The ultimate goal of this work is to develop the capability to calculate flow noise for three dimensional undersea structures for which analytical approaches are not possible.

3:50

1pSA7. Energy finite element formulation for unbound acoustic domains. Nickolas Vlahopoulos (Univ. of Michigan, 2600 Draper Dr., Ann Arbor, MI 48109, nickvl@umich.edu) and Sergey Medyanik (Michigan Eng. Services, Ann Arbor, MI)

The Energy Finite Element Analysis (EFEA) method has been developed for conducting structural-acoustics simulations for complex vehicles at mid-to-high frequencies where conventional finite element methods are no longer computationally efficient. In applications where exterior heavy fluid loading effects must be included in the model, these effects are currently accounted in the EFEA formulation through an added mass and a radiation damping approach. This is suitable when there is an interest in computing the vibration of the structure and the total radiated power emitted in the exterior fluid. A new formulation for modeling the exterior fluid explicitly with energy finite elements is presented. There are three main components in the new development: deriving the differential equation; solving it numerically using a finite element approach; and introducing infinite finite elements in order to represent the non-reflective conditions at the outer domain of a finite element model. In this presentation, the technical aspects associated with the main elements of the new EFEA formulation for exterior non-reverberant acoustic domains will be discussed. Comparisons with analytical closed form solutions for validating the new developments and their numerical implementation will be presented.

Contributed Paper

4:10

1pSA8. Effectiveness of energy finite element analysis applied to submerged undersea vehicle noise prediction. Michael A. Jandron, Robert M. Koch (Naval Undersea Warfare Ctr., Code 8232, Bldg. 1302, Newport, RI 02841, michael.jandron@navy.mil), Allan F. Bower (School of Eng., Brown Univ., Providence, RI), and Nickolas Vlahopoulos (Dept. of Naval Architecture and Marine Eng., Univ. of Michigan, Ann Arbor, MI)

Traditionally, solving structural-acoustics problems has posed significant computational challenges at very high wavenumbers because of the mesh refinement required. As such, instead of resolving each wavelength, the fundamental goal in Energy Finite Element Analysis (EFEA) is to average the energy over many wavelengths. The envelope of average energy behaves as a slowly varying exponential, which is much easier to solve numerically. EFEA thus computes the average energy over a region which can be used to determine undersea vehicle self- and radiated noise in much the same way as conventional FEA modeling but the size of the mesh does not suffer the same restrictions. In this talk, self-noise model predictions for a representative undersea vehicle are discussed to demonstrate the effectiveness of EFEA. This model includes an explicitly modeled acoustic domain to allow radiated energy to propagate in the acoustic medium and possibly reenter the vehicle structure. Results are validated with dense FEA models and the post-processing algorithms used for this purpose are discussed.

Session 1pSC

Speech Communication: Methods and Models for Speech Communication (Poster Session)

Stephanie Del Tufo, Chair
Dept. of Psychol., Univ. of Connecticut, 406 Babbidge Rd., Storrs, CT 06269-1020

All posters will be on display from 1:00 p.m. to 5:00 p.m, To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 1:00 p.m. to 3.00 p.m. and contributors of even-numbered papers will be at their posters from 3:00 p.m. to 5:00 p.m.

Contributed Papers

1pSC1. Validation of relative fundamental frequency using an aerodynamic estimate of vocal effort. Yu-An S. Lien (Biomedical Eng., Boston Univ., 540 Memorial Dr., Apt. 810, Cambridge, MA 02139, slien@bu.edu), Carolyn M. Michener, and Cara E. Stepp (Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

Clinical assessment of the voice disorder vocal hyperfunction currently relies on subjective interpretations of clinicians, which can be unreliable, necessitating the development of objective measures. One relatively established objective measure of vocal hyperfunction is the ratio of sound pressure level to subglottal pressure (dB SPL/cm H_2O). However, this measure provides unreliable results in individuals with vocal fold lesions, which often accompany vocal hyperfunction. A new acoustic measure, relative fundamental frequency (RFF), defined as the normalization of voiced fundamental frequencies surrounding a voiceless consonant, has also been developed for vocal hyperfunction assessment. Here, RFF was validated against dB SPL/cm H_2O in a cohort of healthy speakers simulating various degrees of vocal effort. Twelve healthy participants (M = 22 years; five males) produced stimuli at five levels of effort (ranging from relaxed to maximally strained). The relationship between dB SPL/cm H_2O and RFF was determined. RFF parameters were able to explain 37–47% (R^2) of the variance seen in the aerodynamic measures when averaged across speakers. For six of the speakers, RFF parameters explained 43–97% of the variance, suggesting that the relationship between RFF and dB SPL/cm H_2O may depend on the strategy for modulating vocal effort employed by speakers.

1pSC2. Independent and interacting effects of sentential context and phonological neighborhood structure in spoken word production. Neal P. Fox, Megan Reilly, and Sheila E. Blumstein (Cognit., Linguistic & Psychol. Sci., Brown Univ., Box 1821, Providence, RI 02912, neal_fox@brown.edu)

Models of speech production typically invoke at least two cognitive systems: a structurally static lexicon and a dynamic, compositional system. A speaker must have access to a network of lexical representations that is vested with certain intrinsically associated information (e.g., words' phonological structure), but typical speech involves sequential, online production of words to construct meaningful sentences. Thus, a complete model of speech communication must overlay the lexicon's static architecture with a dynamic, context-driven system. The present study examines effects of these two systems on spoken words' phonetic realizations. Participants produced monosyllabic words beginning with voiceless stop consonants (e.g., *coat*) that varied in their phonological neighborhood density. Targets were produced aloud after subjects silently read sentence contexts that were either highly predictive (e.g., *She sported her stylish new fur...*) or relatively neutral (e.g., *The artist had trouble drawing the...*). Results show that subjects hyperarticulate (i.e., produce with longer voice-onset times) initial stop consonants of words with denser neighborhoods and words that appear in less predictable contexts. Interestingly, the effect of a word's context on its phonetic realization only arises for words in sparse neighborhoods, supporting a model in which the lexicon's static architecture constrains the influence of dynamic processing during production.

1pSC3. Using criterioň voice familiarity to augment the accuracy of speaker identification in voice lineups. Julien Plante-Hébert and Victor J. Boucher (Laboratoire de Sci. phonétiques, Université de Montréal, Montréal, QC H3C 3J7, Canada, julienph85@hotmail.com)

Voice familiarity is a principal factor underlying the apparent superiority of human-based vs machine-based speaker identification. Our study evaluates the effects of voice familiarity on speaker identification in voice lineups by using a Familiarity Index that considers (1) recency (the time of last spoken contact), (2) duration of spoken contact, and (3) frequency of spoken contact. Three separate voice-lineups were designed each containing ten male voices with one target voice that was more or less familiar to individual listeners (13 per lineup, n = 39 listeners in all). The stimuli consisted in several verbal expressions varying in length, all of which reflected a similar dialect and the voices presented a similar speaking fundamental frequency to within one semitone. The main results showed high rates of correct target voice identification across lineups (>99%) when listeners were presented with voices that were highly familiar in terms of all three indices of recency, duration of contact, and frequency of contact. Secondary results showed that the length of the verbal stimuli had little impact on identification rates beyond a four-syllable string.

1pSC4. Domain of final lowering in spontaneous Japanese. Kikuo Maekawa (Dept. Corpus Studies, Natl. Inst. for Japanese Lang. and Linguist., 10-2 Midori-cho, Tachikawa-shi, Tokyo 190-8561, Japan, kikuo@ninjal.ac.jp)

There are two opposing predictions about the domain of final lowering (FL) in Japanese intonation. One predicts that the domain is the last mora of utterance whereas the other predicts that the domain is much wider (with no clear specification of the domain). X-JToBI annotated part of the Corpus of Spontaneous Japanese (the CSJ-Core, 44 h speech spoken by 201 speakers) was analyzed to determine the domain of FL in Tokyo Japanese. Mean normalized F0 values of the four constituent tones of accented accentual phrases (AP) were compared across utterances differing both in the number of constituting APs (from 1 to 5) and the syntactic strength of sentence- or clause-boundaries (three levels). It turned out that all tones in the last AP of utterance were considerably lowered in all utterances regardless of utterance length. This suggests strongly that the domain of FL is the last AP in Japanese. It also turned out that FL occurred in much wider contexts than hitherto believed. FL was observed not only in typical sentence boundaries but also in various syntactic clauses, and there was correlation between the degree of FL and the strength of syntactic boundaries.

1pSC5. Within-session stability of acoustic features of conversational and clear speech. Sarah H. Ferguson, Shae D. Morgan, Lydia R. Rogers (Commun. Sci. and Disord., Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112, sarah.ferguson@hsc.utah.edu), and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., East Lansing, Utah)

In several planned studies, talkers will perform a set of speech tasks several times in each recording session. For example, one study will have talkers perform a set of tasks four times: in quiet, in two levels of noise, and in reverberation. It is unknown, however, whether or how much the acoustic details of speech are affected by simple repetition of the speech task. Over the course of a recording session, talkers' speech might become less careful due to fatigue, boredom, familiarity with the speech materials, or some combination of these factors. Such repetition effects could offset the effects of speaking style instructions given at the beginning of a session. The present study assessed speech acoustic changes over four repetitions of a speech production task set performed under conversational or clear speech instructions. Sixteen talkers performed three speech production tasks (a passage, a list of sentences, and a picture description) four times in each condition. The two speaking styles were recorded in separate test sessions. Several acoustic features relevant to clear speech (e.g., vowel space and speaking rate) will be compared between the first and fourth repetition in each speaking style as well as between the styles.

1pSC6. A noise robust Arabic speech recognition system based on the echo state network. Abdulrahman Alalshekmubarak and Leslie S. Smith (Computing Sci. and Mathematics, Univ. of Stirling, Stirling, Scotland FK9 4LA, United Kingdom, l.s.smith@cs.stir.ac.uk)

A major challenge in the field of automated speech recognition (ASR) lies in designing noise-resilient systems. These systems are crucial for real-world applications where high levels of noise tend to be present. We introduce a noise robust system based on a recently developed approach to training a recurrent neural network (RNN), namely, the echo state network (ESN). To evaluate the performance of the proposed system, we used our recently released public Arabic dataset that contains a total of about 10 000 examples of 20 isolated words spoken by 50 speakers. Different feature extraction methods considered in this study include mel-frequency cepstral coefficients (MFCCs), perceptual linear prediction (PLP) and RASTA- perceptual linear prediction. These extracted features were fed to the ESN and the result was compared with a baseline hidden Markov model (HMM), so that six models were compared in total. These models were trained on clean data and then tested on unseen data with different levels and types of noise. ESN models outperformed HMM models under almost all the feature extraction methods, noise levels, and noise types. The best performance was obtained by the model that combined RASTA-PLP with ESN.

1pSC7. Sub-cultural acoustical perversions and deviations: Where sensory rituals inform transcendence. Judy E. Battaglia (Loyola Marymount Univ., One LMU Dr., Los Angeles, CA 91501, judy.battaglia@lmu.edu), Roxanne A. Banuelos (Commun. Studies, Westwood College, Los Angeles, CA), Ashley L. Cordes (Loyola Marymount Univ., Northridge, CA), and Amanda McRaven (Theatre Arts, California State Univ., Northridge, Northridge, Northridge, CA)

This study investigated the perversions (to the normative culture that the subculture comes in contact with) and perceptions of sound in various subcultures, specifically in the Barona Band of Mission Indians and tribal subsets of rave culture. Ethnographic approaches lead us to understand alteration of sounds as a means to create a space of transcendence and liminality. Specifically, we studied the Barona Band of Mission Indians and their use of silence and sound swallowing as acoustics. Similarly, at various rave cultures, acoustic overstimulation was perceived as a key ingredient to achieving similar forms of collective effervescence. This subculture engaged in discourse likening acoustics to a variety of creative metaphors. Both sound isolation and the unique sense-making processes and speech codes of the subcultures were key to the study. Above all, we investigated the alteration of acoustics within a socio-spiritual context. In an increasingly postmodern and globalized world of tenuous connections and synchronous communication we argued for a need to put a focus on community-based subcultures in order to understand our perceptions of sound in the contemporary cultural moment.

1pSC8. Individual differences in the perception of fundamental frequency scaling in American English speech. Nanette Veilleux (Simmons College, 300 The Fenway, Boston, MA 02139, veilleux@simmons.edu), Jon Barnes, Alejna Brugos (Boston Univ., Boston, MA), and Stefanie Shattuck-Hufnagel (Massachusetts Inst. of Technol., Cambridge, MA)

Although most participants (N = 62) in an F0 scaling experiment judged open syllables (day) as higher in pitch than closed syllable tokens (dane, dave) with the same F0 contour, a subset did not. Results indicate that, in general, listeners perceptually discount F0 over coda regions when judging overall F0 level, and the degree of discount is related to the (lack of) sonority in the coda: day tokens are judged significantly higher than dane tokens which are judged significantly higher than dave tokens with the same F0 contour (dane-dave $p < 0.001$, dane-day $p < 0.01$). However, individual differences are observed: ten listeners showed no significant differences in the perception of F0 levels between the three types of tokens. On the other hand, a contrasting subset of ten subjects demonstrated highly significant differences ($p < 0.001$). The remaining 42 subjects behaved similarly to the entire subject pool with only slightly less significant differences between dane and day F0 level judgments ($p < 0.05$). Therefore, for about 16% of subjects, the F0 over the coda is not discounted in judging F0 level. These individual responses in F0 scaling perception mirror differences found in the Frequency Following Response (e.g., [1]) and could indicate individual differences in F0 processing.

1pSC9. Characteristics of speech production in a post-glossectomy speaker with a free flap: A case study. Xinhui Zhou (Elec. Eng., Univ. of Maryland, College Park, 4325 Rowalt Dr., Apt. 101, College Park, MD 20740, zxinhui2001@gmail.com), Jonghye Woo, Maureen Stone (Dept. of Neural and Pain Sci. and Orthodontics, Univ. of Maryland Dental School, Baltimore, MD), and Carol Espy-Wilson (Elec. Eng., Univ. of Maryland, College Park, College Park, MD)

It is unclear in glossectomy whether a flap will improve or impair speech, at least in the moderate sized (T2) tongue tumors. To gain some insights into this question, we studied the speech production of a post-glossectomy speaker, who had a T2 tumor surgically removed from the left side of his tongue and, closed with a radial forearm free flap (RFFF). Our acoustic analysis showed that this speaker had a significantly smaller vowel space and a significantly higher center of gravity in "sh", but not in "s", compared with the averages of normal controls or post-glossectomy speakers with primary closures. Based on cine and tagged magnetic resonance (MR) images, we analyzed the vocal tract shapes of two vowels and two fricatives and studied the tongue motion in transition of phonemes on this speaker and two controls. We will compare the vocal tract models between the flap patient and the controls. [This study was supported by NIH R01CA133015.]

1pSC10. The scope of boundary lengthening as a function of lexical stress and pitch accent. Argyro Katsika (Haskins Labs., 300 George St., New Haven, CT 06511, katsika.argyro@gmail.com), Jelena Krivokapić (Dept. of Linguist, Univ. of Michigan, Ann Arbor, MI), Christine Mooshammer (Inst. of German Lang. and Linguist, Humboldt Univ. of Berlin, Berlin, Germany), Mark Tiede (Haskins Labs., New Haven, CT), and Louis Goldstein (Dept. of Linguist, Univ. of Southern California, Los Angeles, CA)

Although the phenomenon of boundary lengthening is well established, the scope of the effect and its interaction with prominence is not well understood. It is known that phrase-final prominence is a determining factor. However, it is unclear whether it is lexical stress or pitch accent that drives the effect, and whether the affected domain is continuous or discontinuous. An electromagnetic articulometer (EMA) study of five speakers of Greek was conducted to examine the effect of (1) boundary (word and IP), (2) stress (ultima, penult, or antepenult), and (3) prominence (accented and de-accented) on the duration of phrase-final word articulatory events. In both accented and de-accented conditions, lengthening affected events that immediately preceded the boundary in stress-final words, but was initiated earlier in words with non-final stress. The affected domain was continuous. The stress effect could also be observed in pausing behavior: pauses following phrase-final words were realized with specific vocal tract configurations, and the articulatory movements forming them were longer when stress was final than when non-final. Based on these results, a theoretical account of

1p MON. PM

boundaries is proposed within the Articulatory Phonology framework, with implications for a cross-linguistic model of prosody. [Work supported by NIH.]

1pSC11. Secondary gestures for consonants: An electromagnetic articulometer study of troughs. Hosung Nam, Christine H. Shadle, Mark Tiede (Haskins Labs., 300 George St. Ste. 901, New Haven, CT 06511, nam@haskins.yale.edu), Elliot Saltzman (Physical Therapy and Athletic Training, Boston Univ., Boston, MA), and Douglas H. Whalen (Speech-Language-Hearing Sci., City Univ. of New York, New York, NY)

Troughs, defined as a discontinuity in anticipatory coarticulation (Perkell 1968), have been shown to occur in the tongue body (TB) in /ipi/ and lips in /usu/. In V_1CV_1 contexts articulators retract from their vowel target during C when not active during C. We investigate the trough phenomenon by testing three hypotheses: (1) high intraoral air pressure during the consonant pushes TB down; (2) absence of vowel articulator activation during C in VCV causes TB to move to the neutral position; (3) the trough is a gesture in itself. Electromagnetic articulometer (EMA) data of the tongue, lips, and jaw were tracked for a trained phonetician producing non-words and real words, varying vowel context and the place, manner, and consonant length. For H1, /iCi/ productions with C={/p,f,m/} and three C durations were studied. For H2, /h/ and /ʔ/ in /iCi/ were examined. Troughs were found in all the labials but not in /h, ʔ/, disproving hypothesis 2; trough magnitude decreased in the order: /f/ > /p/ > /m/, disproving hypothesis 1. Trough magnitude increased with C duration, consistent with H3. Results from asymmetric vowel contexts also support H3, and will be discussed within the context of Articulatory Phonology. [Work supported by NIH.]

1pSC12. Principal component analysis of formant transitions. Jamie Wennblom-Byrne, Peter J. Watson (Speech-Language-Hearing Sci., Univ. of Minnesota—Twin Cities, 164 Pillsbury Dr., Shevlin 115, Minneapolis, MN 55455, wennb012@umn.edu), He Huang, and Erwin Montgomery (Greenville Neuromodulation Ctr., Greenville, PA)

Formant transitions represent dynamic articulation processes and are useful to describe differences of articulation (e.g., normal and disordered speakers). Our method, considers the formant frequency change over time as a vector. We first reduce this vector to a single value which is plotted multi-dimensional space. Each dimension represents a data point in time and the position of the point of each dimension is the value of the formant frequency. Once the vector has been converted to a single point in the multidimensional space, principal component analysis is applied to reduce the dimensionality. These dimensions can then be applied to multi-variant statistical analysis. Additionally, the eigenvectors in each principal component can be used to plot the differences between sets of formant transitions through time. The eigenvectors are first weighted by calculating the Pythagorean distance in the multidimensional space of the first set of principle components and then each eigenvector is multiplied by the percentage of the variance explained by that principle component. Then, the eigenvectors are plotted and superimposed over the formant transitions. The change of height along the length of the curve represents areas along the transition of maximum and minimum differences between sets of transitions.

1pSC13. Phonetics exercises using the Alvin experiment-control software. James Hillenbrand (Western Michigan Univ., 1903 W Michigan Ave., Kalamazoo, MI 49008, james.hillenbrand@wmich.edu)

A collection of computer exercises was developed for use in teaching phonetic transcription to students taking introductory coursework in phonetics. The exercises were developed using Alvin, a software package for the design and online control of behavioral experiments [Hillenbrand and Gayvert, J. Speech, Lang., Hearing Res. **48**, 45–60 (2005)]. The main goal was to allow students to work independently on the routine drill that is needed to learn sound-symbol associations. For example, for a vowel transcription exercise, students hear naturally spoken /hVd/ syllables and are asked to choose among twelve buttons labeled with phonetic symbols. Feedback is provided on each trial, and students have the option of repeating any trials with incorrect responses. Also included are word/phrase transcription exercises in which students hear an utterance and are asked to provide a phonetic transcription. Correct transcriptions are provided following each trial, and a summary of the student's performance is displayed at the end of the exercise. Reverse transcription exercises are also included in which a phonetic transcription is displayed and the student's job is to enter the word or phrase in ordinary orthography.

1pSC14. Determining functional units of tongue motion from magnetic resonance imaging. Jonghye Woo (Dept. of Neural and Pain Sci., Univ. of Maryland, Baltimore, 650 W. Baltimore St., 8 South, Baltimore, MD 21201, jschant@gmail.com), Fangxu Xing (Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD), Maureen Stone (Dept. of Neural and Pain Sci., Univ. of Maryland, Baltimore, Baltimore, MD), and Jerry L. Prince (Dept. of Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD)

The human tongue produces oromotor behaviors such as speaking, swallowing, and breathing, which are executed by deforming local functional units using complex muscular array. Therefore, identifying functional units and understanding the mechanisms of coupling among them in relation to the underlying anatomical structures can aid significantly in the understanding of normal motor control and the development of clinical diagnoses and surgical procedures. Magnetic resonance imaging (MRI) has been widely used to observe detailed structures in the vocal tract and to measure internal tissue motion of the tongue. This work aims at determining the functional units from tagged MRI and muscle maps are extracted from high-resolution MRI. A non-negative matrix factorization method with a sparsity constraint is utilized to extract an activation map for each tissue point using a set of motion quantities extracted from tagged MRI including information from point trajectories (i.e., displacement, angle, and curvature) and strain. The activation map is then used to determine the coherent region using spectral clustering, revealing functional units and their relations to the underlying muscles. We test our algorithm on simple protrusion and speech tasks, demonstrating that the proposed algorithm can determine the correlated patterns of the tissue point tracking trajectories and strain.

1pSC15. Towards the computation of a variability index in orofacial muscle activity patterns during speech production using performance verification studies. Shonda Bernadin (Elec. and Comput. Eng., Florida A&M Univ.-Florida State Univ. College of Eng., 2525 Pottsdamer St., Tallahassee, FL 32310, bernadin@eng.fsu.edu), Megan Macpherson (Commun. Disord., Florida State Univ., Tallahassee, FL), Itiel Agramonte, and Tejal Udhan (Elec. and Comput. Eng., Florida A&M Univ.-Florida State Univ. College of Eng., Tallahassee, FL)

In previous work (Wohlert and Smith, 2002), it was determined that variability in children's speech production is reflected in upper lip muscle activity using electromyographic (EMG) data across repetitions of a phrase. Later studies (MacPherson and Smith, 2013) showed that a lip aperture variability index, which represents the difference in upper lip displacement and lower lip displacement, can also be used as a reliable variability measure from EMG data to determine the effects of multiple repetitions of the same utterance on speech motor production. This study is an extension of previous work and examines the orofacial muscle activity patterns (i.e., EMG data) during speech production in efforts to quantify EMG variability. This information can yield significant insights into how well the speech motor system is functioning in different groups of speakers (e.g., healthy young adults vs. healthy older adults). Initially for this study, kinematic measures of articulatory variability in healthy young adults and healthy older adults were verified using previously developed customized, interactive MATLAB programming. The analysis provided the platform for studying the quantification of variability in EMG data. The results of these performance verification analyses are described in this paper.

1pSC16. Tracking four dimensional muscle mechanics from high-resolution and tagged magnetic resonance imaging. Fangxu Xing (Elec. and Comput. Eng., Johns Hopkins Univ., 3400 N Charles St., Baltimore, MD 21218, fxing1@jhu.edu), Jonghye Woo, Joseph K. Ziemba, Maureen Stone (Neural and Pain Sci., Univ. of Maryland, Baltimore, MD), and Jerry L. Prince (Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD)

Assessment of tongue muscle mechanics during speech helps interpret clinical observations and provides data that can predict optimal surgical outcomes. Magnetic resonance imaging (MRI) is a non-invasive method for imaging the tongue that provides information about anatomy and motion. In

this work, we aim to develop a pipeline to track 4D (3D space with time) muscle mechanics in order to measure motion similarities and differences in normal and glossectomy speakers. The pipeline comprises of several modules including super-resolution volume reconstruction of high-resolution MRI (hMRI) and cine-MRI, deformable registration of hMRI with cine-MRI to establish muscle correspondences, tracking tissue points using incompressible motion estimation algorithm (IDEA) from tagged MRI, followed by calculation of muscle mechanics including displacement, rotation, elongation, etc. IDEA is the 3D motion estimated from harmonic phase (HARP) motion that is obtained from tagged MRI. The proposed pipeline was evaluated on five subjects including both normal and glossectomy speakers, yielding accurate tracking results as visually assessed. In addition, we were able to differentiate normal and abnormal muscle mechanics, potentially providing invaluable information for interpreting clinical observations and predicting surgical outcomes.

1pSC17. Accelerometric correlates of nasalized speech in children. Lenny A. Varghese (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215, lennyv@bu.edu), Joseph O. Mendoza (Dept. of Biomedical Eng., Boston Univ., Boston, MA), Maia N. Braden (Dept. of Surgery, American Family Children's Hospital, Madison, WI), and Cara E. Stepp (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., Boston, MA)

Instrumentation-based scoring methods can be used to supplement auditory assessments, providing a more objective assessment of voice quality. One such metric that can be used to assess resonance disorders, the Horii Oral Nasal Coupling Index (HONC), has been shown to successfully separate nasal from nonnasal utterances, but has not been extensively studied in children. We have previously found that using low-pass filtered versions of the nasal accelerometer and microphone signal used in its computation can reduce variability due to vowel placement that is traditionally reported for these scores, and sought to determine whether this reduction in variability would extend to children's speech. We obtained nasal acceleration and speech signals from 26 children, aged 4–9, during the production of various consonant-vowel-consonant tokens and running speech with controlled vowel and consonant loading. HONC scores were compared using broadband and low-frequency portions of these signals. It was found that applying a low-frequency filter reduced the variability of the HONC scores due to vowel type relative to when broadband signals were used, and the scores could discriminate vowels produced in nasal and non-nasal contexts in children with high accuracy. These results demonstrate the potential of HONC scores as an aid in rehabilitating hypernasal speech.

1pSC18. Tense/lax discrimination in articulatory vowel space: Evidence from electromagnetic articulography. Sonya Mehta, Jun Wang, and William F. Katz (Commun. Sci. and Disord., Univ. of Texas at Dallas, 1966 Inwood Rd., Dallas, TX 75235, naya@utdallas.edu)

A kinematic experiment investigated tongue movement for vowels in American English with the aim of developing spatial targets for use by subjects in real-time interactive learning/remediation environments. A 3D EMA system (AG501, Carstens Medezinelektronic, GmbH) was used to track the position of the tongue (tongue tip, TT, and tongue back, TB) and lips (upper lip, UL, and lower lip, LL) of adult subjects producing the vowels of American English (12 monophthongs and 3 diphthongs) in the consonant environment /bVb/. Here, we asked which of the tongue sensors yields the best spatial separation of the corner vowels /i/, /u/, /æ/ and /ɑ/, and whether vowels that are acoustically close in formant space (e.g., the pairs /i/-/I/ and /e/-/ɛ/) are distinguishable by tongue sensor position. Data were recorded for ten talkers. Points taken at the vowel midpoint of each production were plotted to determine the spatial separation (Euclidean distance) between each vowel region (centroid). Preliminary results suggest TB sensors yield the most discriminable patterns for corner vowels. Tense/lax pairs show smaller differences, as predicted by acoustic theory. For most vowel contrasts examined, patterns of spatial separation appeared sufficient for real-time feedback applications.

1pSC19. Gradient prosodic boundary perception and recursion in syntactic disambiguation. Gregg A. Castellucci and Dolly Goldenberg (Linguist, Yale Univ., Dow Hall (370 Temple St.), Rm. 204, New Haven, CT, gregg.castellucci@yale.edu)

Most research examining prosodic structure has made the assumption that prosodic boundaries are categorical and discrete, and all instances of a specific category are linguistically equivalent (Snedeker and Casserly, 2010). The empirical evidence for this notion is the finding that certain within-category differences do not cause changes in attachment ambiguity (Carlson *et al.*, 2001, experiment 2). However, several studies have shown that prosodic boundaries which arguably instantiate the same category are not only produced differently (Ladd, 1988, 2008; Krivokapić and Byrd, 2012), but listeners are also sensitive to variations in production within categories (Wagner and Crivellaro, 2010; Krivokapić and Byrd, 2012). Such findings suggest that, contrary to traditional assumptions, a certain degree of recursion may exist in prosodic structure. Our study further examines this controversial concept, as we test whether listeners are able to differentiate various meanings of coordinative structures using gradient strengths of a single boundary type while holding all other variables constant. If prosodic structure is non-recursive and phonetic gradiency within a certain boundary category is not linguistically meaningful listeners should not be able to perform this task. However, listeners in this study were able to successfully differentiate coordinative structures based on within-category boundary strength alone.

1pSC20. The path from trilling to frication: Synthesizing Puerto Rican Spanish velar fricative realizations of apical trills. Mairym Llorens (Linguist, Univ. of Southern California, 2130 Park Grove Ave., Los Angeles, CA 90007, llorensm@usc.edu)

Puerto Rican Spanish (PRS) speakers variably produce velar fricative realizations of etymological apical trills. One way to understand this alternation is by attributing it to the two types of trill realization having different gestural goals in PRS. Alternatively, both types of realization may be the result of invariant gestural goals that give rise to distinct attractor states. This study tests whether invariant articulator tasks specified for PRS apical trills can result in successfully synthesized trill and fricative realizations. Comparative measures present in the acoustics of tokens of both types of realization as recorded by native PRS speakers serve as landmarks to determine the success of the articulator model used to produce the synthesized versions. Manipulation of tongue tip gestural stiffness on the one hand and constriction degree on the other provide a way to test which parameter values best capture a bi-goal articulator task. While these findings do not argue against the possibility that PRS velar fricative realizations arise from gestural goals that differ from their apical trill counterparts, they show that this possibility is not the only candidate account for the alternation exhibited in PRS.

1pSC21. Automatic analysis of emotional intonation in Brazilian Portuguese. Waldemar F. Netto, Daniel O. Peres, Marcus M. Martins (Letras Clóssicas e Vernóculas, Univ. of Sao Paulo, Rua Etiópia, 81, Sao Paulo, Sao Paulo 03122020, Brazil, danielperes@usp.br), and Maressa F. Vieira (Letras, Faculdade Sudoeste Paulista, Avaré, Brazil)

This study analyzes emotional speech (anger, sadness, and neutral) in Brazilian Portuguese by testing 13 acoustical parameters, which were automatically processed by the software ExProsodia. The data analyzed consisted of excerpts of spontaneous speech of male and female subjects which were collected on Internet. The software selected Intonation Units (IU) based on fundamental frequency (FO), duration, and intensity information. A pilot test was carried out and its result showed that five out of 13 parameters analyzed were statically significant: skewness of fundamental frequency—sF0; coefficient of variation of fundamental frequency—cvF0; difference between medium tone and IU value—dMTIU; standard deviation of intonation unit—sdIU; and coefficient of variation of positive focus/emphasis values (above MT)—cvposF/E. A two-way ANOVA test showed a significant result. A Tukey's HSD test pointed out to differences between anger and other emotions in both genres. A significant difference between genres was found only in sadness. A Cluster test was performed based on the acoustic parameters taken into consideration and showed that only anger (male and female) could be described based on these parameters. Regarding gender differences, only sadness and neutral speech could be clustered. The results indicate that the automatic analysis done by ExProsodia can be a reliable way to analyze emotional speech produced by male and female subjects.

1pSC22. Development of a parametric basis for vocal tract area function representation from a large speech production database. Adam Lammert and Shrikanth Narayanan (Signal Anal. & Interpretation Lab., Univ. of Southern California, 3740 McClintock Ave., Rm. 400, Los Angeles, CA 90089, lammert@usc.edu)

Parametric representations of the vocal tract area function have long been of interest in speech production research because of their close relationship with speech acoustics. Many representations are possible depending on which qualities are considered desirable. For instance, several representations have been developed which trade articulatory interpretability for low-dimensionality and other useful mathematical properties, such as orthogonality. It has not been well-established whether these abstract representations are related to inherent constraints on vocal tract deformation (e.g., articulatory subspaces), or whether these representations can achieve high accuracy with a small number of parameters when applied to area functions from large amounts of continuous speech. The present study approaches these issues by extracting a data-driven basis representation for vocal tract area functions, and subsequently demonstrating the relationship between that basis and previously proposed spatial Fourier bases. Analysis was performed on more than a quarter-million individual area functions extracted from the USC-TIMIT database using region-of-interest analysis. Results suggest that a spatial Fourier basis with only four harmonics provides over 90% accuracy, and that one additional basis function corresponding to labial aperture increases representation accuracy to near 100%.

1pSC23. Misparsing coarticulation. John Kingston (Linguist, Univ. of Massachusetts, 150 Hicks Way, 226 South College, Amherst, MA 01003-9274, jkingston@linguist.umass.edu), Alexandra Jesse (Psych., Univ. of Massachusetts, Amherst, MA), Amanda Rysling (Linguist, Univ. of Massachusetts, Amherst, MA), and Robert Moura (Psych., Univ. of Massachusetts, Amherst, MA)

Considerable evidence shows that listeners often successfully compensate for coarticulation, and parse the speech signal's acoustic properties into their articulatory sources. Our experiments show pervasive misparsing of the acoustic effects of anticipatory coarticulation. Listeners categorized more of a front-back vowel continuum as back before [p] than [t]. A following [p] causes F2 and F3 to fall at the end of preceding vowels, and these listeners treated these falls as information that the vowel was back because F2 and F3 are also lower in back vowels. In experiment 1, misparsing was more extensive in a lax front-back continuum [ɛ-ʌ] than in a tense one [e-o], but the lax vowels were also shorter. In experiment 2, the durations of lax and tense vowels were equalized, and the steady-state:transition ratios were varied from the naturally occurring 70:30 to 50:50 and 30:70. Listeners still misparsed consonantal formant transitions more for lax than tense vowels, likely because the acoustic differences between [ɛ] and [ʌ] are smaller than those between [e] and [o]. They also misparsed more as transitions lengthened relative to the steady-state, which suggests that they did not adjust for differences in speaking rate that would affect the steady-state:transition ratios.

1pSC24. Nonspeech-nonspeech auditory contrast. Tracy Krug and John Kingston (Linguist Dept., Univ. of Massachusetts, Amherst, MA 01003-9274, tkrug@umass.edu)

Many perceptual adjustments for a target speech sound's context have two explanations: listeners could compensate for coarticulation or perceive the target as contrasting auditorily with its context. Only auditory contrast can explain listeners' similar adjustments for non-speech contexts that acoustically resemble the original speech contexts. This experiment tests auditory contrast further by replacing the target sound with non-speech, too. Stimuli consisted of a sequence of two equal ERB fraction spaced tone complexes separated by a short gap. Either the same higher or lower 1 ERB-wide band of tones was amplified in both complexes, creating HH and LL stimuli, or a band 1 ERB higher or lower, creating LH and HL stimuli. Tones were amplified to create HL, LH, HH, and LL stimuli near lower, intermediate, and higher frequencies (1800, 2300, and 2800 Hz). If frequency change within a stimulus evokes auditory contrast, then HL-LH stimulus pairs should be more discriminable than HH-LL pairs, where frequency only changes between stimuli. Listeners discriminated HL-LH stimulus pairs better than HH-LL pairs in all three frequency ranges, as predicted if the complexes contrast within but not between stimuli. They also discriminated both kinds of pairs worse at higher than lower frequencies.

1pSC25. Machine classification versus human judgment of misarticulated pediatric speech. Madhavi Ratnagiri, Kyoko Nagao (Ctr. for Pediatric Auditory and Speech Sci., Alfred I. duPont Hospital for Children, Wilmington, DE), Sara Krieger (Dept. of Communicative Disord., West Chester Univ., West Chester, PA), Linda Vallino, and H. Timothy Bunnell (Ctr. for Pediatric Auditory and Speech Sci., Alfred I. duPont Hospital for Children, 1701 Rockland Rd, Wilmington, DE 19807, bunnell@asel.udel.edu)

In this study, we compared machine classification to expert human judgments of segments produced by children with speech sound disorders and typically developing children. Stimuli were presented for classification in a five-alternative forced-choice paradigm with the target segment presented either in isolation or in word context. The phonemes we were mainly interested in were /s/ and /ʃ/, but instances of /θ/, /t/ and /f/ were also included among the stimuli since they could be confused with some error productions. Experienced speech language pathologists identified the target phonemes in both context conditions and their judgments were compared to the results of an HMM classifier that was trained on the speech of typically developing children. For segments presented in context, percentage agreement between the machine classifier and the consensus human classification was 95.83%. For segments presented in isolation, agreement dropped to 87.29%. Thus, the machine classifier responded more like human listeners responding to segments in word context rather than in isolation. An analysis of the differences between specific stimuli responsible for the lower agreement with isolated segments will be presented.

1pSC26. Using lateral sensors in flesh-point tracking of /l/ and /ɹ/ in American English. William F. Katz, Jun Wang, and Sonya Mehta (Commun. Sci. and Disord., Univ. of Texas at Dallas, 1966 Inwood Rd., Dallas, TX 75235, wkatz@utdallas.edu)

Whereas electromagnetic articulography (EMA) studies commonly use a midsagittal sensor array to record articulatory patterns, higher spatial imaging data (e.g., MRI, ultrasound) suggest that some sounds, such as /ɹ/ and /l/, involve tongue concave/convex shape differences that are more effectively measured along a coronal axis. We therefore explored the use of a lateral sensor in the EMA measurement of liquid consonants. Ten adult subjects produced /l/ and /ɹ/ in /ɑCɑ/, /iCi/, and /uCu/ syllables while seated in a 3D electromagnetic articulography system (AG501, Carstens Medezinelektronic, GmbH). Speech movement was tracked for tongue sensors (tongue tip, TT, tongue lateral, TL, and tongue body, TB) and lips (upper lip, UL, and lower lip, LL). Preliminary results suggest that the TL sensor, taken together with TT and TB, provide an improved characterization of American English liquid consonants. Further results will be discussed in the context of developing methods to optimize real-time speech training and speech rehabilitation systems.

1pSC27. PRAATR: An architecture for controlling the phonetics software "PRAAT" with the R programming language. Aaron L. Albin (Dept. of Linguist, Indiana Univ., Memorial Hall 322, 1021 E 3rd St, Bloomington, IN 47405-7005, aaalbin@indiana.edu)

An increasing number of researchers are using the R programming language (http://www.r-project.org/) for the visualization and statistical modeling of acoustic-phonetic data. However, R's digital signal processing capabilities are still limited compared to free-standing phonetics software such as PRAAT(http://www.fon.hum.uva.nl/praat/). As such, it is typical to extract the acoustic measurements in PRAAT, export the data to a textfile, and then import this file into R for analysis. This process of manually shuttling data from one program to the other slows down and complicates the analysis workflow. The present work reports on a software architecture ("PRAATR")

designed to overcome this inefficiency. Each of its R functions sends a shell command to the operating system that invokes the command-line form of PRAAT with an associated PRAAT script that imports a file, applies a PRAAT-command to it, and then either brings the output directly into R or exports the output as a textfile. Since all arguments are passed from R to PRAAT, the full functionality of the original PRAAT command is available inside R, making it possible to conduct the entire analysis within a single environment. Moreover, with the combined power of these two programs, many new analyses become possible. Further information on PRAATR can be found at the author's website.

1pSC28. Phonetic realization of Japanese vowel devoicing. Kuniko Nielsen (Linguist, Oakland Univ., 320 O'Dowd Hall, Rochester, MI 48309-4401, nielsen@oakland.edu)

Despite the widely used term "vowel devoicing", previous studies have reported that the phonetic realization of Japanese devoicing ranges from devoiced vowel to complete vowel deletion, and is often described as deleted or reduced as opposed to "devoiced" (Beckman, 1982; Keating and Huffman, 1984; Tsuchida, 1997; Kondo, 1997; Maekawa and Kikuchi, 2005, Ogasawara and Warner, 2009). The current study presents an acoustic investigation of phonetic implementation of Japanese vowel devoicing to examine the relative likelihood of vowel devoicing, reduction, and deletion. Twenty-four native speakers of Tokyo Japanese produced 80 words containing high vowels in single devoicing environments. The data revealed that the majority of vowel devoicing tokens (>95%) were phonetically realized as vowel deletions, with no trace of devoiced vowels (= energy excited at frequencies of vowel formants with an aspiration source). Given this result, vowel "deletion" may better characterize the phonetic implementation of Japanese devoicing. Theoretical implications of this finding for Japanese phonological structure will be discussed, including an account that considers mora as the fundamental phonological unit at which devoicing takes place and subsequently views devoicing as a loss of mora sonority.

1pSC29. Gestural coordination of the velum in singing can be different from coordination in speech. Reed Blaylock, Adam Lammert, Louis Goldstein, and Shrikanth Narayanan (Univ. of Southern California, 1150 W 29th St. Apt. 4, Los Angeles, CA 90007, reed.blaylock@gmail.com)

Professional singers are trained to maximize vowel duration and minimize consonant interference, while still maintaining intelligibility. The mechanisms by which they do this, however, are unclear. A deeper understanding of the gestural mechanisms utilized during professional-quality singing could be useful to train singers more effectively. It has been well-established that nasal sounds in syllable coda position have longer and larger velum motion in formal speech as compared to casual speech. The hypothesis of this study was that coda velum gestures would be temporally extended proportional to the duration of the entire syllable. To test this, real-time magnetic resonance (rtMR) images of trained soprano singers were used to track the motion of the velum. The boundaries of the vocal tract were mapped to a semi-polar grid, and the velocity of the velum at every frame of its movement was calculated. An important result shows that operatic singing has minimal change in velum gesture duration compared to speech, while a musical theater style can have velum lowering for the duration of a word. Observed gestural kinematics and their implications for speech are discussed from within the framework and perspective of Articulatory Phonology.

1pSC30. A comparison study of emotional speech articulations using the principal component analysis method. Sungbok Lee, Jangwon Kim, and Shrikanth Narayanan (Elec. Eng., Univ. of Southern California, 3601 Watt Way, GFS-301, Los Angeles, CA 90089, sungbokl@usc.edu)

In this study, we will investigate differences in the tongue movements across emotions by utilizing the principal component analysis (PCA), which enables to detect major and minor variations in the entire tongue surface movements under different speaking conditions. For the purpose of the study, we analyze an acted emotional speech production corpus collected from one actor and two actresses using an electromagnetic articulography (EMA). Discrete emotion types considered in this study are anger, sadness, and happiness as well as neutral one as reference. Specifically, we will investigate the number of principal components that are needed to capture emotional variations, and the differences in tongue shaping across different emotion types. Outcome of the study would provide supplementary information to the previous PCA-based production studies which have mainly focused on normal, or neutral, speech articulation. Major principal components that are found in the study also can be utilized as a basis by which effective but compact articulatory correlates (i.e., component scores) can be derived for a further investigation of emotional speech production such as an interaction between articulatory kinematics and prosodic modulations of pitch and loudness patterns, which is another important motivation of the study.

1pSC31. The effect of contextual mismatches on lexical activation of phonetic variants. Kevin B. McGowan (Linguist, Stanford Univ., Margaret Jacks Hall, Bldg. 460, Stanford, Michigan 94305, kbmcgowan@stanford.edu) and Meghan Sumner (Linguist, Stanford Univ., Stanford, CA)

Listeners' everyday experience of speech is highly varied and much of this variation is richly informative. We routinely make accurate use of phonetic cues in fast speech, for example, that would be ambiguous if decontextualized. In an influential study, Andruski *et al.* (1994) showed that a word like "king" facilitates the recognition of a semantically-related target like "queen" better with a canonically long VOT on the initial [k] than when portions of this VOT are spliced out. These and similar findings are cited as evidence for idealized or canonical lexical representations–a surprising result given the rarity of these forms in listeners' experience. In a series of semantic priming experiments, we test the hypothesis that such findings are attributable not to a benefit for the canonical variant but to a mismatch between the manipulated variants and the contextual phonetic frame in which they are presented. The short VOT variants facilitate recognition equally robustly as their long VOT counterparts when each is presented in an appropriate phonetic frame. Our results make it difficult to argue for a canonical bias in perception but, instead, connect such findings to converging evidence from the coarticulation literature showing that coarticulation facilitates perception.

Session 1pSP

Signal Processing in Acoustics: Sonar and Underwater Acoustic Communications

Paul J. Gendron, Chair
ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747

Contributed Papers

1:30

1pSP1. A derivation and discussion of a mutual information lower bound for dynamic acoustic response functions under a Gauss Markov law. Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, pgendron@umassd.edu)

A derivation and discussion of a mutual information lower bound for broadband acoustic channels under a Gauss-Markov assumption is provided. This lower bound is associated with dynamic broadband acoustic response functions that obey a Gauss Markov law. Considered here is the difference of the mutual information between source and receive signals given the acoustic response and the mutual information between acoustic response and receive signal given the source signal. The derivation illuminates the features of this mutual information and permits identifying two key information loss mechanisms. The first is an added self noise term proportional to signal power that is only associated with the innovation variance of the Gauss Markov response. The second term also increases with the innovation variance but is associated with the uncertainty in the state of the channel operator at the initial time of signaling. This later variance is identified as a forward-backward channel estimation error. Platform motion effects and their impact on these two loss mechanisms are discussed and the structure of an effective signal to noise ratio (eSNR) ceiling at high SNR are identified.

1:45

1pSP2. Blind deconvolution of extended duration underwater signals. Jane H. Kim and David R. Dowling (Mech. Eng., Univ. of Michigan, 2010 Lay Auto Lab, Ann Arbor, MI 48109, janehjki@umich.edu)

Synthetic time reversal (STR) is a passive blind deconvolution technique for estimating the original source signal and the source-to-array impulse responses in an unknown multipath sound channel. Previous investigations of STR involved relatively short duration chirp signals (50 ms) where the assumption of a static underwater sound channel was acceptable. However, the static-channel assumption is inadequate for longer duration underwater acoustic communication signals lasting one or more seconds. Here, wave-driven changes in the ocean's surface shape and water-column sound speed lead to significant temporal variations in the source-to-array impulse responses. This presentation describes an effort to accommodate such temporal variations by decomposing a long duration signal into smaller overlapping pieces, applying STR to each piece, and then stitching the resulting sequence of signal estimates together to blindly recover the original long-duration signal. Simulations of a one-second-duration synthetic signal propagating in a static underwater sound channel to a 16-element vertical array are used to determine how the technique's performance depends on the duration and overlap of the signal pieces in relation to the signal's bit rate and the sound channel's time-delay spread. Extension of this effort to recordings from dynamic sound channels is anticipated. [Sponsored by the Office of Naval Research.]

2:00

1pSP3. A simulation based study of the effective signal to noise ratio of broadband underwater acoustic communications between moving platforms. Shrey Joshi and Paul J. Gendron (ECE Dept., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, sjoshi@umassd.edu)

Characterization of the effective signal to noise ratio (eSNR) of broadband underwater acoustic environments as a function of platform motion provides a practical and useful predictive metric of the adversity of an acoustic communication channel. Presented here is an analysis of estimated and actual eSNR associated with diverse moving platform scenarios for the practical case of an acoustic response function modeled as a Gauss-Markov process. A ray model of the arrival structure of various refractive environments along with a set of canonical motion scenarios provides a space of time-varying channel operators that captures the bulk effects of platform motion. Considered here are various superpositions of elliptical and translational motion effects. It is demonstrated that the eSNR exhibits both a low SNR gap and a high SNR ceiling relative to the actual received SNR (rSNR). These results give insight into useful source power levels and provide a correspondence between coherence degradation and vehicle speed and acceleration rates. These model based results lend credence to previous observations of eSNR made from high frequency broadband underwater acoustic observations [2008 NRL Review, pp. 123–125], [J. Acoust. Soc. Am. **130**(4), (2011)].

2:15

1pSP4. Development of a high frequency underwater acoustic communication modem. Brady Paradis (Elec. Eng., UMass Dartmouth, 37 Horizon Dr., Tiverton, Rhode Island 02878, bparadis@umassd.edu), Corey Bachand (BTech Acoust. LLC, Fall River, MA), Paul Gendron, and David A. Brown (Elec. Eng., UMass Dartmouth, Dartmouth, MA)

The development of a high frequency (300 kHz) underwater acoustic communication modem is reported. The transmit and data acquisition components have been designed for QPSK waveforms, intended to be computationally simple, power efficient, and cost effective. Applicability to other types of waveforms will be discussed. The design incorporates a digital signal processor, an efficient power amplifier, an analog to digital converter, and broadband transducers. An electrical equivalent circuit of the transducer was constructed as an accurate test load for bench-top measurements. Modeled results are compared to in-water test tank measurements with broadband transducers. The system is intended to serve as a development and demonstration platform to test novel waveforms and serve as a communication system on unmanned underwater vehicles (UUVs).

2:30

1pSP5. The generalized sinusoidal frequency modulated active sonar waveform ambiguity function: Theoretical and experimental results. David A. Hague and John Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, david.a.hague@gmail.com)

The generalized sinusoidal FM (GSFM) waveform modifies the sinusoidal FM (SFM) waveform to use an instantaneous frequency (IF) function that resembles a FM chirp waveform and as a result of this the GSFM possesses a thumbtack ambiguity function (AF) [Hague, Buck, Asilomar (2012)]. This is a drastic improvement over the SFM, which suffers from poor range resolution as its AF contains many ambiguous peaks generated by the periodicity of the SFM's IF. Analyzing the Taylor Series expansion of the GSFM's AF near the origin proves that the GSFM achieves minimal range-Doppler coupling for single target measurements. This minimizes the error in jointly estimating target range and velocity. The GSFM's AF Peak Sidelobe Level (PSL) and Integrated Sidelobe Ratio (ISLR) are comparable to or better than that of other waveforms with a thumbtack AF. Consequently, the target masking effect experienced in multi-target environments is less severe for the GSFM than other thumbtack waveforms. Lastly, test tank experiments demonstrate that the GSFM's desirable AF properties are robust to the signal conditioning necessary for transmission on practical piezoelectric transducers. [Work supported by ONR and the SMART Scholarship Program.]

2:45

1pSP6. An effective Sine-Chirp signal for multi-parameter estimation of underwater acoustic channel. Guang Yang, Wei J. Yin (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Nantong St. 145, Harbin City 150001, China, edit231@163.com), Ming Li (College of Eng., Ocean Univ. of China, Harbin City, China), Rong Z. Pan, and Ling H. Zhou (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin City, China)

An effective Sine-Chirp signal is proposed for the multi-parameter estimation of underwater acoustic channel. The Sine-Chirp signal is made up of a sine signal and a Chirp signal with a fixed time interval. The sine part is adopted to estimate the multi-path number and its corresponding Doppler shift, while the Chirp part is used to estimate the multi-path delay. Furthermore, the Chirp-based decomposition character of the fractional Fourier transform (FRFT) is applied to analyze the Chirp part of the Sine-Chirp signal for the delay estimation, which greatly improves the process efficiency of the Sine-Chirp signal. The simulation results verify the effectiveness of the Sine-Chirp signal.

3:00

1pSP7. Fuzzy statistical normalization constant false alarm rate detector for non-Rayleigh sonar data. Yanwei Xu, Chaohuan Hou, Shefeng Yan, and Jun Li (IOA CAS, Beisihuanxilu No. 21, Beijing 100191, China, xyw@mail.ioa.ac.cn)

A new CFAR detector for non-Rayleigh data based on fuzzy statistical normalization is proposed. The proposed detector carries out the detection with two stages. The first stage of the fuzzy statistical normalization CFAR Processor is background level estimation based on fuzzy statistical normalization. The second stage is signal detection based on the original data and the defuzzification normalized data. Performance comparisons are carried out to validate the superiority of the proposed CFAR detector.

MONDAY AFTERNOON, 5 MAY 2014 — PROVIDENCE 1, 5:15 P.M. TO 6:30 P.M.

Meeting of Accredited Standards Committee (ASC) S1 Acoustics

R. J. Peppin, Chair ASC S1, Chair
5012 Macon Road, Rockville, MD 20852

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, etc. 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 coniunction with the Standards Plenary meeting at 9:15 a_m. on Tuesday, 6 May 2014.

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.

Note: Payment of separate fee required to attend

MONDAY EVENING, 5 MAY 2014 BALLROOM D, 7:00 P.M. TO 9:00 P.M.

Session 1eID

Interdisciplinary: Tutorial Lecture on Sound Reproduction: Science in the Service of Art

Alexander U. Case, Chair
Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854

Invited Paper

7:00

1eID1. Sound reproduction: Science in the service of art. Floyd E. Toole (Harman Int. Industries, Inc., 8500 Balboa Blvd., Northridge, CA 91329, soundnwine@sbcglobal.net)

The vast majority of music we enjoy is generated by loudspeakers of differing pedigree and propagated to our ears through spaces that can mostly be described as acoustically arbitrary. In spite of the obvious huge variations, humans have managed to not only enjoy reproduced music, but sometimes even to exhibit enthusiasm for it. Common acoustical measurements confirm the variations. Are they wrong? In double-blind subjective evaluations of loudspeakers in rooms, listeners exhibit strong and remarkably consistent opinions about the sound quality from loudspeakers. The challenge has been to identify those technical measurements that correlate with the subjective ratings. What is it that these listeners are responding to? A clue: it is not in the specification sheets of most loudspeakers, nor in a secret formula for room design. This tutorial will examine the acoustical properties of loudspeakers (the sound source), rooms (the acoustical conveyance), and listeners (the powerfully perceptive, and adaptable receptor). In some respects, our problems began when we started to make certain kinds of simplistic measurements. Two ears and a brain do not respond to complex sound fields the way an omnidirectional microphone and analyzer do. What our eyes see in these measurements is not always well-matched to what our ears hear.

Session 2aAA

Architectural Acoustics: Uncertainty in Describing Room Acoustics Properties I

Lily M. Wang, Cochair
Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816

Ingo B. Witew, Cochair
Inst. of Techn. Acoust., RWTH Aachen Univ., Neustrasse 50, Aachen 52066, Germany

Chair's Introduction—8:15

Invited Papers

8:20

2aAA1. Review of the role of uncertainties in room acoustics. Ralph T. Muehleisen (Decision and Information Sci., Argonne National Lab., 9700 S. Cass Ave., Bldg. 221, Lemont, IL 60439, rmuehleisen@anl.gov)

While many aspects of room acoustics such as material characterization, acoustic propagation and room interaction, and perception have long been, and continue to be, active areas of room acoustics research, the study of uncertainty in room acoustics has been very limited. Uncertainty pervades the room acoustic problem: there is uncertainty in measurement and characterization of materials, uncertainty in the models used for propagation and room interaction, uncertainty in the measurement of sound within rooms, and uncertainty in perception. Only recently are the standard methods of uncertainty assessment being systematically employed within room acoustics. This paper explains the need for systematic study of uncertainty in room acoustic predictions and review some of the most recent research related to characterizing uncertainty in room acoustics.

8:40

2aAA2. Uncertainty and stochastic computations in outdoor sound propagation. D. Keith Wilson (CRREL, U.S. Army ERDC, 72 Lyme Rd., Hanover, NH 03755-1290, D.Keith.Wilson@usace.army.mil) and Chirs L. Pettit (Aerosp. Eng., U.S. Naval Acad., Annapolis, MD)

Outdoor sound propagation provides an interesting and informative example of uncertainty in acoustics. Predictions are strongly impacted by imperfect knowledge of the atmospheric and ground properties, as well as by random turbulence and unresolved elements of the landscape. This presentation describes the impact of such uncertainties and how they can be quantified with stochastic sampling techniques that are applicable to a wide variety of acoustical problems. Efficient and accurate computational approaches result from simultaneously sampling over frequency, uncertain environmental properties, and random processes. Among the techniques considered are ordinary Monte Carlo and Latin hypercube sampling, importance sampling based on relatively simpler propagation models, and adaptive importance sampling. When uncertainties in the atmospheric and ground properties dominate, importance sampling is found to converge to an accurate estimate with the lowest calculation time. When random turbulent scattering dominates, the sampling method has little impact.

9:00

2aAA3. Bias and reproducibility of sound power test methods. Matthew G. Blevins, Lily M. Wang, and Siu-Kit Lau (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 909 S 70th plz #4, Omaha, NE 68106, mblevins@huskers.unl.edu)

Sound power is a useful quantity in describing the strength of an acoustic source because its value is independent of distance. However, many standardized methods exist for the measurement of sound power and comparison between methods can give rise to discrepancies. An interlaboratory study was designed according to the ISO 5725 series to quantify the bias and reproducibility of three common sound power measurement methods in the HVACR industry: free field method, diffuse field method, and sound intensity method. A loudspeaker sound source was used to generate two test signals: a broadband signal with decreasing 5 dB slope per octave band, and the same broadband signal with discrete frequency tones at 58, 120, 300, and 600 Hz. The objective of the study is to quantify repeatability, reproducibility, laboratory bias, and measurement method bias, as well as investigate the influence of tones. The design of the interlaboratory study and preliminary results will be presented. The ISO 5725 methods used to investigate the sound power measurement methods in this study may be applicable to other room acoustic measurements.

9:20

2aAA4. Uncertainty aspects regarding the input for reverberation time predictions. Margriet Lautenbach (Peutz bv, PO Box 696, Zoetermeer 2700 AR, Netherlands, m.lautenbach@peutz.nl) and Martijn Vercammen (Peutz bv, Mook, Netherlands)

The outcome of a reverberation time prediction cannot be more accurate than the combined accuracy of input parameters. In the current procedures of measuring absorption coefficients and using them in prediction models at least two aspects regarding the accuracy of absorption coefficients are quite underexposed:(1) The precision of the measurement: the reproducibility both within one laboratory as between different laboratories or laboratory conditions; (2) The accuracy of the measurement: the dependency of the absorption coefficient on the measurement procedure in general. The upcoming ISO 354 improves both aspects by means of a more elaborate qualification procedure, the use of a reference absorber combined with a correction procedure, as well as a guidance for the extrapolation of the results to other dimensions. Still, it is interesting to think about the impact of the remaining accuracy. To what extend is an acoustic consultant "at risk" using absorption coefficients, apart from using correct modeling algorithms. Two different, but common cases, can give an idea of the influence of the accuracy, two situations in which the reverberation time heavily depends on one absorption material: a class room with an absorptive ceiling, and a concert hall with absorptive chairs.

9:40

2aAA5. Investigations into the sound absorbing properties of gypsum wall board. Robert Healey (Architectural Eng., Univ. of Kansas, 1241 Tennessee Apt. 3, Lawrence, KS 66044, rwhealey@ku.edu), Kevin Butler (Henderson Engineers, Inc., Lawrence, KS), Ian Patrick, and Sean Quigley (Architectural Eng., Univ. of Kansas, Lawrence, KS)

Gypsum wall board is one of the most common materials encountered in buildings and thus encountered in architectural acoustics. Recent research into gypsum wall board sound absorption has indicated that the material may have significantly less absorption than is usually assumed when employed in certain construction assemblies. This paper examines both traditional mountings and a new method for determining sound absorption recently introduced involving mounting a gypsum board assembly in an opening between two reverberant rooms commonly used for transmission loss testing. Sound absorbing data were obtained with the two-room method along with the traditional measurement method, for several gypsum wall board wall assemblies, and with an attempt to compare the measured results to real world experience.

10:00–10:15 Break

10:15

2aAA6. The effects of uncertain scattering coefficients on the reverberation time. Uwe M. Stephenson (Room Acoust., HafenCity Univ. Hamburg, Nelkenweg 10, Bad Oldesloe 23843, Germany, post@umstephenson.de) and Alexander Pohl (Room Acoust., HafenCity Univ. Hamburg, Hamburg, Germany)

Uncertain scattering coefficients are still a weak point in room acoustical computer simulations and predictions. The definition of the scattering coefficient s, their values as well as the simulation model are uncertain. Not only the roughness of infinite walls but also the edge effect are included in a combined diffusivity coefficient ray tracing models utilize. Most interesting in practical room acoustics is the influence on the uncertainty of the reverberation time (RT), which depends very sensitively on s. However, there is no analytical formula for the RT for only partially diffusely reflecting surfaces. In non-diffuse sound fields, the RT depend on the room shape, the distribution of absorption and especially on the scattering coefficients, too. A crucial example is a long rectangular room with totally absorbing side walls and partially scattering front walls. For this case a semi-analytical approach had been found. These and other typical cases in 2D and 3D have now been investigated numerically by the Sound Particle Simulation Method (SPSM) and the Anisotropic Reverberation Model (ARM). The aim is to find semi-analytically formulae or rules to estimate the uncertainty in the prediction of the RT by the Sabine formula. Is an "equivalent scattering area" a useful concept?

10:35

2aAA7. Uncertainty in scattering coefficient measurements of sintered ceramic tiled surfaces. David T. Bradley (Phys. + Astronomy, Vassar College, 124 Raymond Ave., #745, Poughkeepsie, NY 12604, dabradley@vassar.edu), Rhett Russo (School of Architecture, New Jersey Inst. of Technol., Newark, NJ), Ariana Sharma, and Jacob Adelgren (Phys. + Astronomy, Vassar College, Poughkeepsie, NY)

The reflection of acoustic energy in an enclosed space can sometimes lead to undesirable effects, particularly if the reflection is relatively large in amplitude or delayed in time. To mitigate these effects, surfaces with non-planar geometries, which are known as diffusers, can be employed in an architectural space to improve the acoustical qualities of the space by attenuating these harsh reflections and by producing a more evenly distributed sound field. One of the standardized quantifiers used to characterize the effectiveness of a diffuser is the scattering coefficient, which is defined as the ratio of non-specularly reflected energy to total reflected energy. Measuring the scattering coefficient requires a carefully controlled acoustic testing facility known as a reverberation chamber. These chambers often have attributes that can be difficult to control (e.g., humidity) or that do not adhere to the standardized specifications (e.g., size). The current study explores the uncertainty associated with a series of measurements focused on testing the effectiveness of a new type of diffuser, which has been created using a novel ceramic sintering technique. The scattering coefficients of several of these ceramic surfaces have been measured in three different reverberation chambers with varying results.

10:55

2aAA8. Characterization of the uncertainty and error propagation in sound field diffusion measurements. Jin Yong Jeon, Muhammad Imran, and Hyung Suk Jang (Architectural Eng., Hanyang Univ., 17 Haengdang-dong, Seongdong-gu, Seoul, 133791, South Korea, jyjeon@hanyang.ac.kr)

ISO 3382-1 describes instructions for conducting acoustical measurements in auditoria. This standard defines how to measure and derive general properties of the acoustic conditions. By applying the general instructions in ISO, a modified Np was investigated by counting the number of peaks from room impulse responses in an auditorium to characterize the sound field diffuseness. The accuracy of the parametric value in a space depends on the uncertainties involved in all stages of the measurements and analyzing processes. Uncertainties related to source/receiver characteristics, directional aspects, and computational techniques are considered in computing the modified Np. Therefore, we discuss the extent to which we are uncertain about the modified Np and the factors affecting its accuracy in all processes involved in measurements and computations.

11:15

2aAA9. Uncertainty in acoustic metrics due to spatial variations of the non-diffuse sound field measured in a variable acoustics classroom. Zhao Peng, Matthew G. Blevins, and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska-Lincoln, 1110 S. 67th St., Omaha, NE 68182, zpeng@huskers.unl.edu)

Acoustic metrics are commonly expressed as single numbers in classroom acoustical designs, often neglecting the physical quantity's uncertainty due to the non-diffuse sound field in the seating area. A database of measured monaural and binaural room impulse responses (RIR) was previously gathered from a fully furnished mock-up classroom. Different wall and ceiling absorption configurations were used to alter the mid-frequency reverberation times (RT) in five scenarios between 0.4 and 1.1 s. The middle three RT scenarios were additionally created from two different material configurations. For each material configuration (eight in total), two furniture orientations were utilized. RIRs were measured at 9 to 10 receiver positions for each material/furniture configuration to document the spatial variation in the resulting sound field. Diffuseness has been calculated for each receiver position utilizing the measured RIRs by following Hanyu's (2013) method using normalized decay-canceled impulse responses. Variations in diffuseness and in the assorted acoustic metrics calculated from the measured RIRs are investigated across different receiver positions. These acoustic metrics, pertinent to classroom acoustical designs, include RT, speech transmission index, clarity, and interaural cross-correlation. Means to quantify uncertainty in these metrics due to spatial variation in the non-diffuse sound field will be discussed.

11:35

2aAA10. Uncertainty versus parameter in room acoustics. A case study. Miguel Arana, Abel Arregui, Jorge Machin, and Ricardo San Martin (Phys., Public Univ. of Navarre, Campus de Arrosadia, Pamplona, Navarra 31006, Spain, marana@unavarra.es)

An exhaustive characterization of the auditorium of Baranain (Navarre, Spain) has been carried out. All acoustic parameters (both monaural and binaural) in many seats (96 for monaural and 48 for binaural) have been measured for three source positions on the stage. For acoustic characterization, a countless results can be obtained in accordance (in all cases) with the views of the ISO-3382 for the presentation of the results. The spatial dispersion for each source position and combinations thereof will be shown. Accuracy on the acoustic evaluation of the room will be discussed from a statistical point of view.

TUESDAY MORNING, 6 MAY 2014 — 554 A/B, 8:00 A.M. TO 11:25 A.M.

Session 2aAB

Animal Bioacoustics and Signal Processing in Acoustics: Dynamics of Biosonar: Time-Dependent Adaptations in Transmission, Reception, and Representation in Echolocating Animals II

James A. Simmons, Chair
Neurosci., Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912

Invited Papers

8:00

2aAB1. Characteristics of echolocation system in Japanese house bat, *Pipistrellus abramus*. Hiroshi Riquimaroux (Life and Medical Sci., Doshisha Univ., 1-3 Miyakotani, Tatara, Kyotanabe 610-0321, Japan, hrikimar@mail.doshisha.ac.jp)

Japanese house bats, *Pipistrellus abramus*, emit harmonically structured downward FM sweeps for echolocation where the fundamental frequency changes from 80 to 40 kHz. Previous studies with an on-board telemetry microphone revealed that they conduct echo-amplitude compensation to stabilize amplitude of returning echoes. When they hunt preys in the field, emitted pulses are prolonged, and

the fundamental frequency little change where the terminal fundamental frequency is fixed at about 40 kHz. About 40% of neurons in the inferior colliculus are tuned best to a frequencies range between 35 and 45 kHz, where higher frequency resolution than other frequency ranges is implied and suited for detecting wing beats of insects. However, their audiogram has not yet been known. The present study constructed their audiograms between 4 and 80 kHz by using the auditory brainstem responses evoked by tone pips. Results show that threshold around 40 kHz is lower than other frequencies except for frequency regions around 20 kHz. Findings suggest that Japanese house bats appear to have high sensitivity around 40 kHz, the terminal frequency, and around 20 kHz, corresponding to their communication frequencies. Characteristics of their cochlear microphonics will be discussed. [Work supported by ONR.]

8:20

2aAB2. Adaptive changes in acoustic characteristics of echolocation pulses emitted by Japanese house bats under artificial jamming conditions. Shizuko Hiryu, Eri Takahashi, Kiri Hyomoto, Yoshiaki Watanabe, Hiroshi Riquimaroux, and Tetsuo Ohta (Faculty of Life and Medical Sci., Doshisha Univ., 1-3 Tatara Miyakodani, Kyotanabe 610-0321, Japan, shiryu@mail.doshisha.ac.jp)

The echolocation behavior of *Pipistrellus abramus* during exposure to artificial jamming sounds during flight was investigated. Echolocation pulses were recorded using a telemetry microphone mounted on the bats' backs, and their adaption based on acoustic characteristics of emitted pulses was assessed in terms of jamming-avoidance responses (JAR). In experiment 1, FM jamming sounds mimicking echolocation pulses of *P. abramus* were prepared. All bats showed significant increases in the terminal frequency (TF) by an average of 2.1–4.5 kHz when the TF of the jamming sounds was lower than the bats' own pulses. This frequency shift was not observed using jamming frequencies that overlapped with or were higher than the bats' own pulses. These findings suggest that JAR in *P. abramus*are sensitive to the TF of jamming pulses and that the bats' response pattern was dependent on the slight difference in stimulus frequency. In experiment 2, when bats were repeatedly exposed to a band-limited noise, the bats in flight more frequently emitted pulses during silent periods between jamming sounds. These results demonstrate that bats could rapidly adjust their vocalized frequency and emission timing to avoid frequency and temporal overlap with jamming sound even during flight. [Research supported by JSPS.]

Contributed Papers

8:40

2aAB3. Dynamics of biosonar signals of free swimming dolphins searching for bottom targets. 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 (Navy Marine Mammal Program, Spa War System Ctr., San Diego, CA), and Patrick W. Moore (National Marine Mammal Foundation, San Diego, CA)

Measuring on-axis biosonar signals from a free swimming dolphin performing a sonar task is extremely difficult without having a special device that the animals carry. A bite-plate device which had a hydrophone directly in front of the dolphin at a fixed location along the beam axis of the biosonar beam was constructed as a part of the Navy Marine Mammal Research Program in San Diego. A data acquisition and storage unit was a part of the Biosonar Measurement Toolbox (BMT) and hung below the bite plate. The outgoing signal was measured by the omnidirectional hydrophone while two disk hydrophones measured the echoes. The device was used with two Atlantic bottlenose dolphins (*Tursiops truncatus*) as the animals conducted biosonar searches for specific objects on the ocean bottom. The outgoing signals were parameterized into the following, peak-to-peak source levels, source energy flux density, center frequency, rms bandwidth, rms duration, and interclick intervals. The parameters were analyzed with both a cluster analysis and a principle component analysis to determine the grouping of the parameters and the relationship between parameters. Both dolphins used different biosonar search strategy in solving the problem and their biosonar signals reflect the difference in strategy.

8:55

2aAB4. Origin of the off-axis double-pulse in an echolocating bottlenose dolphin. Dorian S. Houser, Brian Branstetter, Jason Mulsow, Patrick Moore (National Marine Mammal Foundation, 2240 Shelter Island Dr., San Diego, CA 92106, dorian.houser@nmmfoundation.org), and James J. Finneran (SSC Pacific, San Diego, CA)

It has been proposed that two pairs of phonic lips might be utilized by a dolphin to adaptively manipulate the frequency content and directionality of the echolocation beam in response to acquired target information. The presence of two pulses appearing off-axis of the echolocation beam and separated in time has been proposed as evidence supporting this hypothesis. An array containing 35 hydrophones was used to measure the beam pattern of a bottlenose dolphin performing a phantom echo-change detection task. Simulated target ranges varied from 2.5 to 80 m and clicks were measured at 5–10° resolution from +/150°. At recording angles beyond +/30°, the click appeared as two distinct pulses that declined in amplitude and distorted as the off-axis angle increased. A simple model utilizing the time difference of arrival for the two pulses was used to compare the direct source-receiver path to one of two source-reflector-receiver paths. Assuming a range of constant sound speeds, distances traveled were compared to a CT scan of the same animal to predict anatomical regions potentially contributing to the second pulse. Results suggest that the second pulse is due to reflections from internal structures of the dolphin head and not a second sound source.

9:10

2aAB5. Can you hear me now? Sensitive hearing and limited variation in wild beluga whales (*Delphinapterus leucas*). T. Aran Mooney (Biology Dept., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, amooney@whoi.edu), Manuel Castellote (National Marine Mammal Lab., Alaska Fisheries Sci. Ctr., National Marine Fisheries Service, Seattle, WA), Lori Quakenbush (Arctic Marine Mammal Program, Alaska Dept. of Fish and Game, Fairbanks, AK), Roderick Hobbs (National Marine Mammal Lab., Alaska Fisheries Sci. Ctr., National Marine Fisheries Service, Seattle, WA), Caroline Goertz (Alaska SeaLife Ctr., Seward, AK), and Eric Gaglione (Georgia Aquarium, Atlanta, GA)

Odontocetes use sound for vital biological functions such as foraging, navigation, and communication, and hearing is considered their primary sensory modality. Yet, hearing abilities within wild species or populations are essentially unknown. Hearing sensitivities and variations can greatly influence biosonar signals and performance. Here we present the hearing abilities of seven wild beluga whales. Data were collected in a capture-release event in Bristol Bay, AK. The goal was to establish the mean audiogram and variation within a subset of animals, defining what wild belugas hear and examining how sound-sensitivities may differ between individuals. Hearing was measured using auditory evoked potentials, from 2 to 150 kHz. All belugas tested could hear up to at least 128 kHz, and two heard up to 150 kHz, exceptionally high for belugas. Regions of "best" hearing (<60 dB) were found between 22 and 100 kHz, showing generally broad sensitivity. Greatest variation (>40 dB) was at regions of best sensitivity and the highest frequencies. This substantial variation was less than that of bottlenose dolphins suggesting differences between populations and the need to collect comparative data on new species. While generally sensitive, the hearing variability suggests that multiple analyses better describe the maximum sensitivity and population variance for odontocetes.

9:25

2aAB6. Echolocation strategy for multiple target-preys by foraging bats investigated by field measurement and mathematical modeling. Emyo Fujioka (FIRST, Aihara Innovative Mathematical Modelling Project, JST, 4-6-1-Cw601 Komaba, Meguro-ku, Tokyo, Japan, Tokyo 153-8505, Japan, emyo.fujioka@gmail.com), Ikkyu Aihara (Brain Sci. Inst., RIKEN, Saitama, Japan), Shotaro Watanabe (Faculty of Life and Medical Sci., Doshisha Univ., Kyoto, Japan), Miwa Sumiya, Shizuko Hiryu, Yoshiaki Watanabe, Hiroshi Riquimaroux (Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan), and Kazuyuki Aihara (Inst. of Industrial Sci., The Univ. of Tokyo, Tokyo, Japan)

Using a super microphone-array system, 3-D flight paths of echolocating Japanese house bats, *Pipistrellus abramus*, across whole foraging area were successfully measured in the field, together with the directional aim of their sonar beams. Sonar sounds were sometimes rapidly alternated between its direction and other one or two particular directions during search phase. Especially, when the bats consecutively captured multiple insect preys, their emissions were directed toward not only the current target prey and also the next target. These suggest that the bats process multiple echo streams by time-sharing manner and plan the flight path to efficiently capture multiple target preys. In order to examine whether the bats select the efficient flight path to consecutively capture multiple targets, the bats' 3-D flight behavior while approaching two target preys was then modeled. The modeling analysis suggested that the echolocating bats select their flight paths to easily direct their sonar beams toward both targets. [This research was supported by the Aihara Project, the FIRST program from JSPS, initiated by CSTP, a Grant-in-Aid for Young Scientists (A) of JSPS, and the Murata Science Foundation.]

9:40

2aAB7. Eigenbeam analysis of the diversity in bat biosonar beampatterns. Philip Caspers, Alexander Leonessa, and Rolf Mueller (Mech. Eng., Virginia Tech, 1110 Washington St., SW, MC 0917, Blacksburg, VA 24061, pcaspers@vt.edu)

A quantitative analysis of the interspecific variability in the biosonar beampatterns of bats has been performed on a data set that consisted of 267 emission and reception beampatterns from 98 different species. The beampatterns were aligned using a pairwise optimization framework defined by a rotation for which a cost function is minimized. The cost function was defined by a p-norm computed over all direction and summed across a discrete set of evenly sampled frequencies. For a representative subset of beampatterns, it was found that all pairwise alignments between beampatterns result in a global minimum that fell near the plane bisecting the mean direction of each beampattern and containing the origin. Following alignment, the average beampattern was found to consist of a single lobe that narrowed with increasing frequency. Variability around the average beampattern was analyzed using principle component analysis (PCA) that resulted in "eigenbeams": The first three "eigenbeams" were found to control the beamwidth of the beampattern across frequency while higher rank eigenbeams accounted for symmetry breaks and changes in lobe direction. Reception and emission beampattern could be differentiated based on their PCA scores using only a small number of eigenbeams.

9:55

2aAB8. Influence of mouth opening and gape angle on the transmitted signals of big brown bats (*Eptesicus fuscus*). Laura N. Kloepper (Dept. of Neurosci., Brown Univ., 185 Meeting St. Box GL-N, Providence, RI 02912, laura_kloepper@brown.edu), Jason Gaudette (Adv. Acoust. Div., Naval Undersea Warfare Ctr., Providence, RI), James Simmons (Neurosci., Brown Univ., Prov, RI), and John R. Buck (Elec. and Comput. Eng., Univ. of Massachusetts Dartmouth, Dartmouth, MA)

The big brown bat (*Eptesicus fuscus*) produces echolocation sounds in its larynx and emits them through its mouth. The bat is presumed to change the directionality of the emitted echolocation beams by modifying its mouth opening width. We analyzed infrared video sampled at 240 fps synchronized to ultrasonic recordings from a Knowles Electret microphone sampled at 192 kHz. Mouth angles for each emitted echolocation pulse were calculated offline and compared to the pulse's time-frequency characteristics. Our results indicate that the mouth influences both the amplitude and spectral characteristics of the emitted pulse.

10:10–10:25 Break

10:25

2aAB9. Target shape perception and clutter rejection use the same mechanism in bat sonar. Michaela Warnecke (Neurosci., Brown Univ., 185 Meeting St., Providence, RI 02912, michaela_warnecke@brown.edu) and James A. Simmons (Neurosensing and Bionavigation Res. Ctr., Doshisha Univ., Providence, Rhode Island)

Big brown bats (*Eptesicus*) emit multiple-harmonic FM sounds (FM1, FM2) and exploit the relative weakening of higher harmonics in lowpass echoes from the surrounding scene to suppress clutter by defocusing of wideband images. Only echoes from a frontally located targets arrive as unfiltered, focused images. Experiments using electronically generated echoes show that lowpass filtering of masking echoes causes clutter masking to disappear. Lowpass filtering induces amplitude-latency trading, which retards response times at higher frequencies in clutter echoes relative to lower frequencies. Introducing countervailing changes in presentation-times of higher frequencies in electronically generated clutter echoes restores masking. In the big brown bat's inferior colliculus, FM sounds mimicking broadcasts and echoes evoke ~1 spike per sound at each neuron's best frequency; however, amplitude tuning is very broad. To exploit their high acuity for detecting coherence or non-coherence of echo responses, bats work in the latency domain instead, removing background objects through deliberate imposition of response de-synchronization and concomitant inattention on undesired clutter. Overall, the results indicate that big brown bats use neuronal response timing for virtually all auditory computations of echo delay, including clutter rejection. This use of active perceptual processes in biosonar instead of conventional sonar processes opens a new view toward biomimetic design.

10:40

2aAB10. Acoustic tracking of bats in clutter environments using microphone arrays. Ikuo Matsuo (Dept. of Information Sci., Tohoku Gakuin Univ., Tenjinzawa 2-1-1, Izumi-ku, Sendai 9813193, Japan, matsuo@cs.tohoku-gakuin.ac.jp), Alyssa Wheeler, Laura Kloepper, Jason Gaudette, and James A. Simmons (Dept. of Neurosci., Brown Univ., Providence, RI)

The big brown bat, *Eptesicus fuscus*, uses echolocation for foraging and orientation. Bats can change the echolocation calls dependent on the environments. Therefore, it is necessary to clarify the changes of acoustic characteristics of these calls. In this study, the flight path of the bat were tracked by computing the time differences of arrivals (TDOA) at the microphone array system in the flight room. The acoustic patterns of echolocation calls could be calculated from the measured call data at each microphone by compensating the spread and absorption loss. The head aim and beam pattern at each harmonics were computed from these acoustic patterns of echolocation calls. It was examined whether these acoustics beam patterns were dependent on clutter environment, that is, density of chains. [This research was supported by ONR, NSF, and JST, CREST.]

10:55

2aAB11. Bats (*Tadarida brasiliensis*) jam conspecifics in food competition. Aaron J. Corcoran (Biology, Univ. of Maryland, 4309 Rowalt Dr. #201, College Park, MD 20740, aaron.j.corcoran@gmail.com) and William E. Conner (Biology, Wake Forest Univ., Winston Salem, NC)

We here describe field experiments testing whether bats adaptively produce sounds to interfere with ("jam") the echolocation of other bats. Visual observations, low-light videography, and ultrasound recording with microphone arrays allowing reconstruction of bat flight paths were used to document interactions between Mexican free-tailed bats (*Tadarida brasiliensis*) at two foraging locations in southern Arizona and New Mexico. We tested three sets of predictions based on the jamming hypothesis and two competing hypotheses—cooperative foraging and food patch defense. Bats produced putative jamming calls (termed sinFM calls) that overlapped

2a TUE. AM

temporally and spectrally with "feeding buzz" calls made by conspecifics when attacking insects. Bat capture success decreased by 400% when sinFM calls were present compared to when they were absent. Behavioral sequences consisted of two or more bats sequentially making feeding buzzes within a restricted area while another bats made sinFM calls. After making sinFM calls bats frequently turned toward where the other bat made its feeding buzz and then made a buzz of its own. Together, the results support the hypothesis that bats jam conspecifics in extended bouts of food competition. This is the first known case of echolocating animals adaptively jamming conspecifics.

11:10

2aAB12. Temporal dynamics of echolocation during clutter navigation in *Eptesicus fuscus*. Alyssa Wheeler, Laura Kloepper (Neurosci., Brown Univ., 189 Meeting St., Box GL-N, Providence, RI 02903, Alyssa_Wheeler@Brown.edu), Ikuo Matsuo (Information Sci., Tokohu Gakuin Univ., Sendai, Japan), and James A. Simmons (Neurosci., Brown Univ., Providence, RI)

The echolocating big brown bat, Eptesicus fuscus, must both avoid background objects that it may collide with such as vegetation or clutter, and identify insect prey targets. When bats fly through clutter, they emit groups of echolocation sounds in rapid succession called strobe groups. Here, we investigate the limitations of strobe grouping for flight guidance during clutter navigation. We hypothesized that as clutter conditions become extreme, bats will be no longer able to use strobe groups to navigate. We exposed bats to our laboratory flight room cluttered with plastic chains—obstacles that are acoustically similar to vegetation. Bats flew down a corridor 1.0 m wide, 0.7 m wide, or 0.4 m wide, and echolocation sounds were recorded with a 22-microphone array. We used a means clustering analysis to quantify inter-pulse intervals (IPIs) as belonging within a strobe group or between strobe groups. We found that bats made significantly more sounds per flight as the corridor width decreased. Strobe groups in the wide condition contained longer within-strobe IPI, while strobe groups in the narrowest condition had the shortest. We also investigated how many sounds per strobe group were used. These results show a relationship between clutter density and the temporal structure of echolocation calls.

TUESDAY MORNING, 6 MAY 2014 BALLROOM E, 8:00 A.M. TO 11:30 A.M.

Session 2aBA

Biomedical Acoustics: Brain Therapy and Imaging

Yun Jing, Cochair
Mech. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus Box 7910, Raleigh, NC 27695

Gregory T. Clement, Cochair
Cleveland Clinic, 9500 Euclid Ave., Cleveland, OH 44195-0001

Invited Papers

8:00

2aBA1. Recent advancement in transcranial brain imaging using photoacoustic computed tomography. Mark Anastasio, Kenji Mitsuhashi, Chao Huang, Robert Schoonover, Konstantin Maslov, Alejandro Garcia-Uribe, and Lihong V. Wang (Washington Univ. in St. Louis, One Brookings Dr., St. Louis, MO 63130, anastasio@wustl.edu)

Photoacoustic computed tomography (PACT) holds great promise for transcranial brain imaging. However, the strong reflection, scattering, and attenuation of acoustic waves by the skull present significant challenges for image reconstruction. In this talk, we will review our recent progress on transcranial PACT image reconstruction. Our contributions include the following: (1) development of a methodology to establish a discrete transcranial PACT image model by use of adjunct X-ray CT data; (2) development of image reconstruction methods that can compensate for speed-of-sound and density variations within the skull; and (3) a detailed investigation of the role of shear waves in transcranial PACT. Computer-simulated and experimental data are employed to demonstrate the feasibility of transcranial PACT.

8:20

2aBA2. Passive mapping of acoustic sources within the human skull cavity with a hemispherical sparse array using computed tomography-based aberration corrections. Ryan M. Jones (Medical Biophys., Univ. of Toronto, 2075 Bayview Ave., Focused Ultrasound Lab (C713), Toronto, ON M4N 3M5, Canada, rmjones@sri.utoronto.ca), Meaghan A. O'Reilly (Physical Sci. Platform, Sunnybrook Res. Inst. , Toronto, ON, Canada), and Kullervo Hynynen (Medical Biophys., Univ. of Toronto, Toronto, ON, Canada)

Traditionally, the use of ultrasound (US) in the brain has been limited by the skull bone, which presents unique challenges for both transcranial therapy and imaging due to its attenuating and aberrating effects, which become more prevalent at higher US frequencies. On transmit, these skull-induced aberrations can be overcome through the use of large-aperture phased array transducers with appropriate driving frequencies, combined with computed tomography (CT)-based bone morphology and numerical models to derive element driving signals which minimize the distortions. Recently, we have demonstrated *in silico* that an analogous approach can be performed during beamforming on receive, to allow for passive acoustic imaging over a large region within the skull cavity [Jones *et al*., *Phys. Med. Biol*. **58**, 4981–5005 (2013)]. We will present preliminary results obtained from applying this technique experimentally with a hemispherical (30 cm diam.) sparse receiver array (128 piezo-ceramic elements, 2.5 mm diam., and 612 kHz center frequency) to image acoustic sources through an *ex vivo* human skullcap. Images produced using non-invasive CT-based skull corrections will be compared with those obtained through an invasive hydrophone-based correction approach, and with images formed without skull-specific corrections. This technique has promising applications in both cavitation-mediated transcranial focused ultrasound therapies, by providing a method for treatment monitoring and control, as well as in ultrasound angiographic imaging of the brain.

8:40

2aBA3. Photoacoustic imaging of brain cortex in rhesus macaques. Xinmai Yang (Mech. Eng., The Univ. of Kansas, 1560 W 15th St., Lawrence, KS 66045, xmyang@ku.edu)

Functional detection in the brain by photoacoustic (PA) imaging has been an area of interest because of its potential to overcome the limitation of the current available techniques. Both small and large animal brains have been studied by PA imaging. Functional detection in primate brains is particularly of interest because of the similarity between non-human primate brain and human brain and the potential for relevance to a wide range of conditions such as stroke and Parkinson's disease. In this presentation, we show the application of PA imaging technique in detecting functional changes in primary motor cortex of awake rhesus monkeys. Strong increases in PA signal amplitude during forelimb movement indicate an increase in activity in primary motor cortex. The results demonstrate that PA imaging can reliably detect primary motor cortex activation associated with forelimb movement in rhesus macaques. The potentials of PA imaging on clinical human brain imaging will also be discussed.

9:00

2aBA4. Transcranial ultrasound hemorrhage detector. Faik C. Meral (Radiology, Brigham and Women's Hospital, 221 Longwood Ave., EBRC 521, Boston, MA 02115, fcmeral@bwh.harvard.edu), Amber B. Bennoui, Joleigh V. Ferro (Chemistry and Phys., Simmons College, Boston, MA), Charles D. Maneval, Mufaddal A. Jafferji, Nicholas J. Giordano (Biomedical Eng., Boston Univ., Boston, MA), Greg T. Clement (Biomedical Eng., Cleveland Clinic Lerner Res. Inst., Cleveland, OH), and Phillip J. White (Radiology, Brigham and Women's Hospital, Boston, MA)

The present state-of-the-art technology in clinical neurosonography requires an acoustic window to achieve a sufficient signal-to-noise ratio (SNR) for detection of intracranial hemorrhage (ICrH). Conventional clinical practices use ultrasound in the range of 2–10 MHz to image and diagnose patients, a frequency range that is too high to produce high-quality transcranial images. This study examined the use of ultrasound frequencies on the order of 0.5 MHz, applied through the skull bone, to detect ICrH. We demonstrated with *ex vivo* human skull specimens, *ex vivo* animal blood, and soft tissue-mimicking phantoms that a single element ultrasonic transducer can likely be used to detect the presence of ICrH immediately adjacent to the skull bone. The lower frequency design of the transducer together with optimized aperture geometry demonstrated efficient signal transmission through the skull bone with a focal depth optimized for intracranial anatomy. The identification of hemorrhage-brain interfaces was demonstrated with bench-top experiments that mimicked the clinical conditions of acute epi- and subdural hemorrhage. Finally, the device limitations were demonstrated through positive predictive value analysis.

9:20

2aBA5. Transcranial ultrasound-optical transmission correlation. Faik C. Meral (Radiology, Brigham and Women's Hospital, 221 Longwood Ave., EBRC 521, Boston, MA 02115, fcmeral@bwh.harvard.edu), Zun Zar Chi Naing, Felicity A. Meyer (Chemistry and Phys., Simmons College, Boston, MA), Mufaddal A. Jafferji, Chanikarn Power (Radiology, Brigham and Women's Hospital, Boston, MA), Greg T. Clement (Biomedical Eng., Cleveland Clinic Lerner Res. Inst., Cleveland, OH), and Phillip J. White (Radiology, Brigham and Women's Hospital, Boston, MA)

Although the transmission of ultrasound (US) through the skull bone has been demonstrated for both therapeutic and imaging applications, the clinical efficacy of certain transcranial US applications remains limited by the highly attenuating properties of skull bone. For those applications, the ability to pre-procedurally determine those areas of the skull that are less attenuating to US could be a tremendous asset for improving the use of US in the brain. To present a possible solution, we have hypothesized that the optical transmission intensity at points across the skull surface can be correlated with the local US transmission efficiency. The demonstration of this correlation would potentially allow for the use of integrated lasers and photodetectors within a HIFU system to create a patient-specific transmission map of the skull. We have statistically investigated the relationship between transmitted optical and US intensities over multiple points across several *ex vivo* human calvaria to demonstrate this correlation. Along with the results of the analysis, preliminary designs to incorporate optical transmission assessment in transcranial HIFU and echoencephalography will be presented.

9:40–10:00 Break

10:00

2aBA6. Transcranial passive cavitation mapping with a linear array: A simulation study with clinical datasets. Costas D. Arvanitis (Radiology, Harvard Med. School, Brigham and Women, 221 Longwood Ave., Boston, MA 02115, cda@bwh.harvard.edu), Gregory Clement (Dept. of Biomedical Eng., Cleveland Clinic Lerner Res. Inst., Cleveland, OH), and Nathan McDannold (Radiology, Harvard Med. School, Brigham and Women, Boston, MA)

While numerous investigations that explore the unique properties of acoustic cavitation have recently demonstrated very promising results for a wide range of applications, there is a need to develop new and noninvasive methods that will lead to (1) deeper understandings of the interactions involved, (2) their optimal use for therapy or diagnosis, and (3) ultimately their translation to the clinics. Toward this aim, we developed finite difference time domain (FDTD) simulations to transcranially map and assess the diverging pressure waves generated by oscillating microbubbles. The skull and brain tissue acoustic properties (density, speed of sound, and absorption) were extracted from clinical CT datasets. Point sources derived by microbubble dynamics models were also incorporated and propagated toward a virtual US array that was used to perform passive acoustic mapping (PAM). The FDTD simulations suggest that the microbubbles' pressure waves propagating through the skull lose 97% of their strength as compared to propagation in water. The simulations also indicated that transcranial PAM is possible with an 80 mm aperture linear array; however, at wider apertures (150 mm), significant aberration was introduced. Incorporation of variable speed of sound to the PAM back-projection algorithm corrects the aberrations and significantly improves the resolution.

10:20

2aBA7. On the use of fast marching methods for transcranial beam focusing. Tianren Wang and Yun Jing (Mech. Eng., North Carolina State Univ., 911 Oval Dr., EB III, Campus box 7910, Raleigh, NC 27695, yjing2@ncsu.edu)

In this talk, we will present our recent studies on the use of fast marching methods for transcranial beam focusing. Three topics will be included: beam focusing for transcranial B-mode imaging, beam focusing for transcranial photoacoustic tomography, and beam focusing of a spherical array for therapy. To correct for the phase aberration from the skull, two critical steps are needed prior to brain imaging or treatment. In the first step, the skull shape and speed of sound are acquired by either CT scans or ultrasound scans. In the second step, fast marching methods are used to compute the phase delay based on the known skull shape and sound speed from the first step, and the computation can be completed in seconds even for 3D problems. The computed phase delays are then used in combination with the conventional delay-and-sum algorithm for generating images. They can also be readily used for transcranial beam focusing for therapeutic purposes. Numerical simulation results will be presented to show the robustness of fast marching methods.

10:40

2aBA8. Drug delivery through the opened blood-brain barrier in mice and non-human primates. Elisa Konofagou, Cherry Chen, Hong Chen, Matthew Downs, Vincent Ferrera, Oluyemi Olumolade, Gesthimani Samiotaki, Tao Sun, Shutao Wang, Shih-Ying Wu (Biomedical Eng., Columbia Univ., 1210 Amsterdam Ave., ET351, New York, NY 10027, ek2191@columbia.edu)

Over five million U.S. men and women suffer from neurodegenerative diseases. Although great progress has been made in recent years toward understanding of these diseases, few effective treatments and no cures are currently available. This is mainly due to the impermeability of the blood-brain barrier (BBB) that allows only 5% of more than 7000 small-molecule drugs available to treat only a tiny fraction of these diseases. Safe and localized opening of the BBB has been proven to present a significant challenge. Focused ultrasound (FUS), in conjunction with microbubbles, remains the sole technique that can induce localized BBB opening noninvasively and regionally. In the past, our group has focused on cavitation monitoring during BBB opening in both mice and non-human primates, assessment of safety and drug efficacy using behavioral testing, delivery of molecules of variant size through the opened BBB, investigation on the role of the microbubble diameter and use of nanodroplets. We will briefly highlight these past findings as well as introduce newer accomplishments such as its role in enhancement of drugs for neuroprotection and neuroregeneration in the treatment of Parkinson's, the use of alternative routes of systemic administration for larger drug dosage, dependence of the BBB opening size on the acoustic pressure, real-time monitoring of the microbubble perfusion of the brain, cavitation prediction of the timeline of BBB opening, and targeted delivery using adeno-associated viruses.

Contributed Papers

11:00

2aBA9. Effects of diffraction on acoustic radiation force produced by sound beams incident on spherical viscoelastic scatterers in tissue. Benjamin C. Treweek (Dept. of Mech. Eng., Univ. of Texas at Austin, 610 West 30th St., Apt. 126, Austin, TX 78705, btreweek@utexas.edu), Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

The theory for acoustic radiation force on a viscoelastic sphere of arbitrary size in tissue was extended at the spring 2013 ASA meeting to account for nonaxisymmetric fields incident on the scatterer [Ilinskii *et al.*, POMA **19**, 045004 (2013)]. The results were presented in a form that permits inclusion of as many spherical harmonics as needed to describe the field structure. At the fall 2013 ASA meeting, it was shown that for spheres having sizes up to about one wavelength, only four or five spherical harmonics are required for convergence of the solution when plane waves are incident on the scatterer. At the present meeting, the model is applied to diffracting sound beams incident on the scatterer. The analysis is based on angular spectrum decomposition of the incident field, expansion of the resulting plane waves in spherical waves, then a Wigner transformation of the latter back into spherical coordinates with polar axis coinciding with the beam axis, and finally integration over solid angle to obtain the spherical wave amplitudes used in the theory. Results are presented for different radiation patterns illustrating dependence of the radiation force both on beamwidth and on wavelength relative to the size of the scatterer.

11:15

2aBA10. A novel device for guiding ventriculostomy with transcranial ultrasound. Faik C. Meral (Radiology, Brigham and Women's Hospital, 221 Longwood Ave., EBRC 521, Boston, MA 02115, fcmeral@bwh.harvard.edu), Michael A. Persaud, Aaron E. Silva, Abhishek Mundra (Biomedical Eng., Boston Univ., Boston, MA), Greg T. Clement (Biomedical Eng., Cleveland Clinic Lerner Res. Inst., Cleveland, OH), Kirby G. Vosburgh, and Phillip J. White (Radiology, Brigham and Women's Hospital, Boston, MA)

A ventriculostomy is often performed to relieve symptoms of emergent hydrocephalus. This involves the placement of an external ventricular drain (EVD) into the cerebral ventricles to remove excess cerebrospinal fluid. Free-handed EVD cannulation results in high rates of misplacement (~50%), leading to an increased risk of iatrogenic complications. Extant technical approaches to improve ventriculostomy guidance are either too complex or inaccurate. We have investigated the possibility of a novel device to guide EVD placement using transcranial ultrasound. The device uses three specifically aligned transducers delivering pulse-echo 0.5-MHz ultrasound through the skull bone to detect and localize the targeted ventricle. It also incorporates a cannula guide that is registered with the ultrasound FOV to integrate guidance with surgery. Results from the design, fabrication, and testing of the prototype device with *ex vivo* human skulls and brain phantoms will be presented.

TUESDAY MORNING, 6 MAY 2014 — 550 A/B, 9:00 A.M. TO 11:50 A.M.

Session 2aEA

Engineering Acoustics: Session in Honor of Kim Benjamin

Thomas R. Howarth, Chair

NAVSEA Div. Newport, 1176 Howell St., B1346 R404A, Newport, RI 02841

Chair's Introduction—9:00

Invited Papers

9:05

2aEA1. Reflections of Kim Benjamin as a student, professional colleague, and personal friend. Peter R. Stepanishen (Ocean Eng., Univ. of RI, Narragansett Bay campus, Sheets Bldg., Narragansett, RI 02882, stepanishen@egr.uri.edu)

Kim was a very special person who I was privileged to know as a student, professional colleague, and a close personal friend. After completing his undergraduate degree in Physics at URI in 1975, he was accepted into the Masters Program in Ocean Engineering at the University of Rhode Island where I served as his advisor and major professor. While working on an NIH grant to develop two-dimensional planar arrays of ultrasonic transducers for medical diagnostic purposes, Kim developed his lifelong passion for acoustic transducers. Some of his early "experiences with transducers" will be shared in the presentation. During this time period, Kim also investigated and developed the use of FFT methods for the backward projection of acoustic fields from planar sources. His early contributions in this area laid the foundation for much of the subsequent acoustical holographic work by others over the following several decades and will be briefly reviewed. In the 1980s and 1990, it was clear that Kim was becoming a "transducer guru" as evidenced by his research, reports, and related ASA presentations. I will share some recollections of Kim from this period and our many "extended lunches" during the last several years.

9:30

2aEA2. An overview of Kim Benjamin's U. S. Navy transducer developments. Thomas R. Howarth (NAVSEA Div. Newport, 1176 Howell St., B1346 R404A, Newport, RI 02841, thomas.howarth@navy.mil)

Kim Benjamin had already enjoyed a distinguished career as an acoustician at the University of Rhode Island and Raytheon Company before joining the U.S. Navy as a civilian scientist in 1995. From 1995 to 2013, Kim focused primarily on advancing 1–3 piezocomposite materials into unique underwater acoustic devices. Among key accomplishments are the following: design of 1–3 piezocomposite-based beam steered parametric mode transducers with integral high-gain receivers; design of parametric mode sub-bottom profiler transducers; development of U. S. Navy calibration transducer standards F82 and F83; use of 1–3 piezocomposites with area shaded electroding to realize a new class of transduction which maintains a constant beamwidth over a two octave bandwidth; novel use of singly and double curved piezocomposites; design and segment demonstration of a cylindrical array module that is coupled linearly to form a towed line array with 3D spatial discrimination; design and fabrication of a 120 element conical octahedral homing array for high speed (>150 knots) applications. He also was involved in the development of thin, low frequency acoustic sources as he designed a complex set of tooling to accomplish mission objectives. This presentation will overview Kim's Navy contributions with design drawings, photographs, and experimental data.

9:55

2aEA3. Novel, broadband piezocomposite transducers for Navy applications. Brian Pazol and Timothy Mayo (Materials Systems Inc., 543 Great Rd., Littleton, MA 01460, bpazol@matsysinc.com)

Piezocomposites (a two phased material consisting of piezoceramic and polymer) are a widely known form of piezoelectric material with proven advantages over conventional monolithic piezoelectric ceramics. Transducers made with piezocomposite are naturally broadband, have high sensitivity, can be easily shaded for sidelobe suppression, and can be conformed to a variety of shapes. Materials Systems Inc. (MSI) utilizes a low cost injection molding technique that allows large areas to be made by tiling ceramic preforms into large areas at a reasonable cost. There is a large design space that allows better transducer optimization by adjusting the matrix material, active material, and the ratio of active to inactive material (volume fraction). Kim Benjamin recognized the benefits of piezocomposite early in its development. Over the years, he developed many novel and high performance broadband piezocomposite transducers. This paper presents a brief overview of several unique piezocomposite transducers that he developed with MSI. These includes a Constant Beamwidth Transducer (CBT), torpedo homing arrays, and towed high power sources for a variety of Navy applications. Several of Kim's novel fabrication techniques and measured transducer performance will be presented.

10:20–10:30 Break

10:30

2aEA4. Constant beamwidth transducers: A tribute to Kim Benjamin. Dehua Huang (NAVSEA Newport, 43 Holliston Ave., Portsmouth, RI 02871, DHHuang@cox.net)

Constant beamwidth transducer (CBT) is a special acoustic transducer, where acoustic beamwidth is independent of frequency, because of its Legendre polynomial normal velocity distribution on the surface of transducer spherical dome. By elegant design of Legendre polynomial normal velocity distribution profile on the transducer radiation surface dome, acoustic sidelobes can also be controlled and eliminated. To achieve Legendre polynomial normal velocity distribution, electrode area shading is one of important techniques to design a practical CBT. in this paper, the CBT designs, shading patterns, size effects, frequency band limit, build of materials, as well as Mr. Kim Benjamin critical contributions in the field toward a U. S. Navy standard CBT transducer at the Underwater Sound Reference DivNpt (USRD) will be summarized.

10:50

2aEA5. Conformal cymbal array: A broadband, wide beamwidth underwater acoustic projector. James Tressler and Brian H. Houston (Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, james.tressler@nrl.navy.mil)

The development of "cymbal"-based acoustic source technology in recent years has been an essential advancement in improving NRL's capability for shallow-water acoustics. The cymbal's unique attributes of high acoustic output from a lightweight, thin profile configurable array is a central technology in target identification programs at NRL. This talk will present a review of cymbal array development over the past ten years. It will start with conventional piezoelectric ceramic (PZT) cymbal elements and end with current research on advanced cymbal designs utilizing new formulations of piezoelectric single crystal materials. In particular, the contributions of Kim Benjamin to the initial development work of the conformal array design will be highlighted. [Work sponsored by the Naval Research Laboratory.]

11:10

2aEA6. Outstanding work of Kim Benjamin on 1–3 composite transducers and acoustic analysis. Kenneth M. Walsh (K&M Eng. Ltd., 51 Bayberry Ln., Middletown, RI 02842, kwalsh4@mindspring.com)

Kim Benjamin was one of the most productive acoustic system researchers that I have known. Kim, myself, and Walter Boober developed acoustic techniques that increased the productivity of the NUWC acoustic facility by a factor of 10. Kim and I produced a number of patents related to the use of 1–3 composite transducers in naval sonars. The last effort with Kim produced a small diameter, broad band transducer that was tested successfully. Kim was a Fellow of the ASA and is greatly missed.

11:30

2aEA7. Transducer cloaking for Kim Benjamin. John L. Butler (Image Acoust., Inc., 97 Elm St., Cohasset, MA 02025, jbutler@imageacoustics.com)

There is an effort to develop metamaterials for cloaking objects in a way that eliminates backscattering and fills in the shadow zone. The development of this cloaking material for spherical and other shapes would inhibit the acoustic detection of objects, such as mines, torpedoes, UUV's, and, ultimately, submarines by a means which would make them invisible to acoustic waves. It has, however, been pointed out that this form of inactive cloaking could cover a target in a way that shields the target from using its own acoustic sonar means for detecting the source, unless the cloaking could be turned off. We address this issue by presenting an active cloaking transducer system which effectively cloaks the target and yet can also be used as an acoustic sonar system. Equivalent circuits and finite element models are used to demonstrate transducer cloaking. This would have been a good transducer project for Kim Benjamin to implement and I believe he would have enjoyed the challenge and developed one of the best cloaking transducer arrays possible.

Session 2aMU

Musical Acoustics: Where Are They Now? Past Student Paper Award Winners Report

James P. Cottingham, Chair
Phys., Coe College, 1220 First Ave., Cedar Rapids, IA 52402

Invited Papers

9:40

2aMU1. Where they are, where they have been, and where they are going. James P. Cottingham (Phys., Coe College, 1220 First Ave., Cedar Rapids, IA 52402, jcotting@coe.edu)

The ASA Best Student Paper Awards competition began in 1997. From 1997 to 2001, one award winner in Musical Acoustics was selected at each meeting, but beginning in 2002, two awards have been given, with first and second prize winners selected. In all there have been 52 award winners in Musical Acoustics, including three who won an award twice. Some are still active in musical acoustics, but many others are now active in other areas of acoustics or in fields outside acoustics altogether. A brief overview will be presented of the history of the competition and past and current interests of those who have been the award winners. The speakers in this session include winners of the award since the last Providence meeting in 2006. Capsule updates on several award winners who are unable to participate in this session will be presented.

10:00

2aMU2. From musical acoustics to outdoor sound and back. Whitney L. Coyle (The Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, wlc5061@psu.edu)

My musical acoustics student paper award was given in 2009 at ASA San Antonio after an NSF summer research experience at Coe College with Dr. James Cottingham. I was a junior at Murray State University in Kentucky studying the clarinet and mathematics. Though I was not the most qualified applicant, lacking the physics background necessary in this field, I had a passion for acoustics and wanted my future to go in this direction. This summer was my introduction to acoustics research and the reason I was able to continue in the field. Since, I have been attending Penn State in the Graduate Program in Acoustics and have earned my Masters in Acoustics focusing on outdoor sound propagation modeling and now, with the help of an NSF-GRFP, I have found my way back to musical acoustics—clarinet acoustics. Since the San Antonio ASA I have attended seven more ASA conferences, been the musical acoustics student council representative and am now the student council chair. I now split my time between Penn State and Marseille, France, at CNRS-LMA. This talk will detail each step along the path that retuned me to musical acoustics and give a look into my current research as well.

10:20

2aMU3. A systems engineering approach to musical acoustics. Nicholas J. Eyring (Dept. of Phys. and Astronomy, Brigham Young Univ., 2353 W 1700 North, Provo, UT 84604, eyringj@gmail.com)

Research in musical acoustics has benefited greatly from advances in technology; however, the ever increasing complexity of measurement systems may lend to inefficient experimental design and procedure. A systems engineering approach to experimentation can improve the process. Systems engineering involves the design of complex, many element, systems that maximize overall performance, considering all elements related in any way to the system, including human efficiency and the characteristics of each of the system's components. This paper explores how a background in the full product development lifecycle of a Raman spectroscopy based measurement instrument allowed for the rapid development of an automated directivity acquisition system (ADAS) used to measure a concert grand piano. Prior experience also assisted in adapting the ADAS to measure musicians when assessing elements like reliability, logistics, work-processes, and optimization methods. An account of how research in musical acoustics provides applicable experience for employment in a non-acoustics industry position will also be given.

10:40

2aMU4. The science of art, the art of science. Rohan Krishnamurthy (Musicology, Eastman School of Music, 544 Sunrise Circle, Kalamazoo, Michigan 49009, rohan.krishnamurthy@rochester.edu)

I will discuss how my passion for the arts and sciences originated in and developed since elementary school, and led me to my present work as a professional percussionist, educator, researcher, and entrepreneur. Throughout high school, I pursued music and science projects in parallel. My dual interests encouraged me to pursue a double major in music and chemistry at Kalamazoo College. My senior thesis introduced me to acoustical research when I worked with Dr. James Cottingham at Coe College to study the acoustics of a new drum tuning system that I invented and patented. I presented my research at ASA 2007 in New Orleans and other research at subsequent

ASA conferences. After finishing college, I pursued a doctorate in musicology at the Eastman School of Music at the University of Rochester. I will discuss specific examples of my past and current interdisciplinary projects, and how my association with the ASA continues to inspire my endeavors.

11:00

2aMU5. Categorization and lexicon in verbal descriptions of violin quality by performers. Charalampos Saitis (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., 555 Sherbrooke Str. West, Montreal, QC H3A 1E3, Canada, charalampos.saitis@mail.mcgill.ca), Claudia Fritz (Lutheries-Acoustique-Musique, Inst. Jean le Rond d'Alembert, Université Pierre et Marie Curie, UMR CNRS 7190, Paris, France), and Gary P. Scavone (Computational Acoust. Modeling Lab., Ctr. for Interdisciplinary Res. in Music Media and Technol., Schulich School of Music, McGill Univ., Montreal, QC, Canada)

This work aimed to explore how violin quality is conceptualized as reflected in spontaneous verbal descriptions by experienced performers collected while playing in a perceptual evaluation experiment. Participants performed a preference ranking task and justified their perceptions in a free verbalization task. Using the constant comparison analysis from grounded theory, a concept map was developed, which can be useful for future studies aimed at assessing violin qualities. A psycholinguistic analysis of the quality-descriptive lexicon used by violinists further revealed a variety of linguistic devices referring to either the sound of a violin or to the violin itself as the cognitive objects. Adjectives for the description of sound characteristics are largely borrowed from four semantic fields related to texture-temperature (smooth vs. rough), action-presence (resonant vs. muted), size-volume (deep vs. flat), and light (dark vs. bright). These semantic fields indicate what type of dimensions may explain the perception of violin timbre, contributing to the area of violin acoustics research as well as to the broader area of timbre research. Some acoustical interpretations are discussed in the context of finding correlations between measurable vibrational properties of a violin and its perceived quality.

11:20

2aMU6. Robotics, free-reed instruments, and naughty birds: Finding the common thread. Eric A. Dieckman (Sonalysts, Inc., 84 Nicoll St., Unit 1, New Haven, CT 06511, eric.dieckman@gmail.com)

Musical acoustics often provides an accessible starting point for undergraduate research in which the basics are learned and applied to interesting problems. Even if students later study other areas of acoustics, the research experience gained is invaluable. This presentation will touch on a number of research projects encountered since the author's 2006 award for investigations of the acoustic behavior of Southeast Asian free-reed mouth organs. These projects come from a wide variety of acoustic disciplines, from nondestructive evaluation and architectural acoustics to benign bird exclusion and acoustic sensors for mobile robots, with the common thread of signal processing providing a focal point.

11:40

2aMU7. An experience-based approach to auditory perception research. Brian B. Monson (Dept. of Newborn Medicine, Brigham and Women's Hospital, Harvard Med. School, 75 Francis St., Boston, MA 02115, bbmonson@email.arizona.edu)

What and how we hear is determined by our past experience. Thus attempting to quantify human experience becomes the challenge for modeling auditory perception. Based on my recent research at the Duke-NUS Graduate Medical School in Singapore, I will discuss some principles that should guide such an approach to study of auditory neuroscience and perception. One crucial principle is to account for the frequency of occurrence of stimulus patterns to which humans have been exposed over phylogeny and ontogeny (an alternative to explaining perception based solely on peripheral auditory physiology). I will discuss the implications of this research approach and include a brief report on my research progress in my new position at Brigham and Women's Hospital.

Session 2aNSa

Noise: Session in Honor of Kenneth Eldred

Louis C. Sutherland, Cochair
lcs-acoustics, 5701 Crestridge Rd., Apt. 243, Rancho Palos Verdes, CA 90275

Paul D. Schomer, Cochair
Schomer and Associates Inc., 2117 Robert Dr., Champaign, IL 61821

Chair's Introduction—8:55

Invited Papers

9:00

2aNSa1. Kenneth McKechnie Eldred—A distinguished noise control engineer I. William W. Lang (Noise Control Foundation, 29 Hornbeck Ridge, Poughkeepsie, NY 12603, lang1ww@gmail.com) and George C. Maling (NAE Member, Harpswell, ME)

Elected in mid-career in 1975 to the U.S. National Academy of Engineering (NAE) "For outstanding accomplishments in noise and vibration control of air, space, and transportation vehicles and in delineating acceptable noise environments for people." His NAE peers recognized him "...as one of the five best noise and vibration control engineers in the country." Ken's first job after graduating from M.I.T. in 1950 with studies of advanced courses in acoustics was as head of the Boston Naval Shipyard's lab working on the reduction of the noise and vibrations of submarine auxiliary equipment. In 1953 on active duty in the U.S. Air Force, he became Chief of the Bio-Acoustics Branch, Wright Air Development Center. In 1957, he moved to California for a career with Western Electro-Acoustic Laboratory and Wyle Laboratories prior to joining Bolt Beranek and Newman. This paper is primarily devoted to his activities in support of INCE/USA, his role in the passage of the Noise Control Act of 1972, and his activities in connection with the Office of Noise Abatement and Control in the U.S. Environmental Protection Agency.

9:20

2aNSa2. Kenneth McKechnie Eldred—A distinguished noise control engineer II. George C. Maling (NAE Member, 60 High Head Rd., Harpswell, ME 04079, maling@alum.mit.edu) and Eric W. Wood (Acentech Inc., Cambridge, MA)

Ken's accomplishments in noise control engineering cover the spectrum from basic engineering research to recommendations for noise control in six areas: measurement of industrial noise; measurement and reduction of structural vibration in space vehicles; noise radiation from jet flow; noise reduction of jets by multiple nozzles and turbofans; vibroacoustic environmental simulation for aerospace vehicles; and community and transportation noise control. Some of these areas are discussed in consulting reports, and others in papers published in the open literature. He was a consultant at Bolt Beranek and Newman from 1973 until 1982 when he formed Ken Eldred Engineering. His publications appeared in the journal NOISE Control, several NOISE-CON and INTER-NOISE Proceedings, *Noise Control Engineering Journal*, and the *Journal of the Acoustical Society of America*. Representative samples of his accomplishments taken from the above sources will be presented.

9:40

2aNSa3. Ken Eldred—Mentor. Richard Potter (Retired, 129 Wilder Dr., Harvest, AL 35749, dickpotter@bellsouth.net)

I met Ken Eldred at Wyle Laboratories in Alabama in 1963 as a young, green engineer shortly after he formed a staff to support NASA's Apollo program. Later, I was joined by other graduates of the Institute for Sound and Vibration Research, Southampton University, England. Ken led us as we undertook exciting and challenging projects. At Wyle, Ken actively mentored us, encouraging us to develop our investigative skills, adapt to new technologies, and write understandable, readable, and useful reports. He encouraged me to produce ideas and he listened to my suggestions, which he carefully reviewed and then, gently, offered corrections and suggested direction. I, and many of us, owe our successful careers to his mentoring. I will describe some particular work and list some who prospered under his leadership. I moved back to England but then, later, returned to the United States and rejoined Ken at Bolt Beranek and Newman in Cambridge. Again he showed interest and encouragement as I expanded my noise work to other aspects of industrial hygiene, later forming my own companies.

10:00

2aNSa4. Tribute to Kenneth McKechnie Eldred. Louis C. Sutherland (lcs-acoustics, 5701 Crestridge Rd., Apt. 243, Rancho Palos Verdes, CA 90275, lou-sutherland@juno.com)

I first met Ken while at the Boeing Co and he was Vice President at the Western Electro-Acoustics Laboratory in Los Angeles. I met him again at Wright Patterson Air Force Base, Dayton, Ohio, where he was Chief of Physical Acoustics under Henning von Gierke. He recruited me to come to Huntsville, Alabama, to join the new branch of Wyle that Marshall Space Flight Center wanted for rocket noise programs supporting NASA's rocket noise programs. To augment this Wyle staff, Ken recruited several outstanding acoustical scientists from Southampton University in the UK, including the late Martin Lowson. More on this in Richard Potter's paper. Ken's Wyle staff worked with the Federal Aviation Administration, the Boeing Co. and Lockheed Aircraft in pursuing the environmentally failed development of the SST. Ken supported the U.S. Environmental Protection Agency Office of Noise Abatement and Control (ONAC) and guided Wyle in their preparation of key documents on Noise Policy. He left Wyle to join Bolt, Beranek and Newman (BBN) and later left BBN to form Ken Eldred Engineering. He was a Fellow of ASA, received the ASA Silver Medal in Noise in 1994, and was active on ASA Standards committees, the National Research Council, the Society of Automotive Engineers, and the National Academy of Sciences. The other speakers will discuss other aspects of Ken's many contributions, including those for the Institute of Noise Control Engineering (INCE).

TUESDAY MORNING, 6 MAY 2014 — 557, 10:35 A.M. TO 12:00 NOON

Session 2aNSb

Noise: Session in Honor of Harvey Hubbard

Louis C. Sutherland, Cochair
lcs-acoustics, 5701 Crestridge Rd., Apt. 243, Rancho Palos Verdes, CA 90275

Paul D. Schomer, Cochair
Schomer and Associates Inc., 2117 Robert Dr., Champaign, IL 61821

Chair's Introduction—10:35

Invited Papers

10:40

2aNSb1. The administrative leadership of Harvey Hubbard. Charles E. Schmid (ASA, 365 Ericksen Ave., Bainbridge Island, WA 98110, ceschmid@att.net)

Anyone who met Harvey Hubbard knew right away that he was a gentleman. His unassuming manner belied the long list of his accomplishments from the time he left a one-room school in Franklin County, Vermont to his death at age 90 in Newport News, Virginia, in 2012. His endeavors were both technical and administrative, but only the latter will be covered in this presentation, leaving the technical topics to other speakers. The focus will first be on his presidency of the Acoustical Society of America (1989–90). His leadership in selecting and hiring the Society's first executive director (which directly impacted this author) was carried out with his usual professional graciousness. A number of other innovations which occurred during his presidency will be described, such as the establishment of an Investment Committee, which has become vital to the Society's economic well-being. He held many managerial positions in other non-profits, for-profits, and governmental organizations, which will be mentioned to fully understand the breadth of his managerial contributions. [The author would like to acknowledge Elaine Moran for tracking down much of the information which will be presented.]

11:00

2aNSb2. Harvey H. Hubbard and his contributions to wind turbine noise (among other things). Kevin P. Shepherd (NASA Langley Res. Ctr., 2 N. Dryden St., Hampton, VA 23681, k.p.shepherd@nasa.gov)

Following his long NASA career devoted primarily to the understanding and reduction of aircraft noise, Harvey Hubbard came out of retirement to pursue an interesting opportunity that was presented by issues concerning the sound from large wind turbines. This paper will attempt to summarize this pioneering work on wind turbine noise, along with some other accomplishments and recollections.

11:20

2aNSb3. Harvey H. Hubbard's contributions to aircraft noise control during his NACA-NASA career. Domenic J. Maglieri (Eagle Aeronautics, Inc., 732 Thimble Shoals Blvd.'Bldg. C 204, Newport News, VA 23606, sonicboomexpert1@verizon.net)

Following his service in the US Army Air Corps during World War ll, Harvey accepted a position at the NACA Langley Memorial Aeronautical Laboratory in 1945. Propeller aircraft dominated the air transport system at that time and Harvey became one of the first to perform research on the noise associated with propellers. In the next decade jet engine powered aircraft made their appearance and they became the focus of a rapidly growing acoustics research program at Langley. Harvey's pioneering experimental noise studies on propellers and jet engines provided significant insight and understanding of these two concerns. Foreseeing that additional research efforts would be required to address the many new aircraft noise issues, including the sonic boom and airport-community noise concerns of the proposed U.S. supersonic transport, he played a key role in getting NASA to expand its acoustic efforts. As a result, Harvey became NASA's technical focal point for all major acoustical activities. This paper will show the acoustic activities that Harvey was involved in and present some highlights of his research on propeller noise, jet noise, and sonic boom.

11:40

2aNSb4. Harvey Hubbard and the Acoustical Society of America oral histories project. Victor Sparrow (Grad. Prog. Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

Harvey Hubbard was an amazing person who contributed substantially to both the profession of acoustics and to the Acoustical Society of America. This short talk is the story of Harvey Hubbard's ASA oral history. It was a great honor to work with Harvey on his oral history. The transcript of the interview is currently available online through the American Institute of Physics' Center for History of Physics and the Niels Bohr Library and Archives. The interviewer for that oral history will recount some of the highlights from Harvey giving that interview and from some of the superlative life experiences he recounted. Everyone is encouraged to participate in the collection of similar oral histories for the preservation of the history of acoustics and the history of the ASA.

TUESDAY MORNING, 6 MAY 2014 — 551 A/B, 7:55 A.M. TO 10:10 A.M.

Session 2aPA

Physical Acoustics: Beyond Basic Crystals: Viscoelastic and Piezoelectric Materials

Julian D. Maynard, Cochair
Phys., Penn State Univ., 104 Davey Lab., Box 231, University Park, PA 16802

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

Chair's Introduction—7:55

Invited Papers

8:00

2aPA1. Resonance ultrasound spectroscopy for studying piezoelectricity and internal friction at elevated temperatures of quartz, langasite, and gallium nitride. Hirotsugu Ogi (Eng. Sci., Graduate School of Eng. Sci., Osaka Univ., Toyonaka, Osaka 560-8531, Japan, ogi@me.es.osaka-u.ac.jp) and Hassel Ledbetter (Mech. Eng., Univ. of Colorado Boulder, Boulder, CO)

Resonance ultrasound spectroscopy (RUS) is a powerful method for measuring elastic constants (Cijkl) of solids. It can be applied to determine the piezoelectric coefficients (eijk) as well, because they also affect the mechanical resonance frequencies. Precise frequency measurements in vacuum and unambiguous mode identification with laser-Doppler interferometry allowed us to determine Cijkl and eijk simultaneously for crystals including alpha quartz, which shows low piezoelectricity. We further developed a noncontact excitation and detection method with antennas through electromagnetic fields and study internal friction and carrier mobility of quartz, langasite, and GaN at elevated temperatures up to ~1200 K. This method is also successfully applied to developing ultrahigh-sensitive biosensors for diagnosis.

8:20

2aPA2. Measuring viscoelasticity of soft tissues using shear and guided waves. Matthew W. Urban, Carolina Amador, Ivan Z. Nenadic, Heng Zhao, Shigao Chen, and James F. Greenleaf (Dept. of Physiol. and Biomedical Eng., Mayo Clinic College of Medicine, 200 First St. SW, Rochester, MN 55905, urban.matthew@mayo.edu)

Elasticity imaging has emerged as a viable clinical tool for assisting in diagnosis of certain diseases such as liver fibrosis and cancer. Many methods have been developed for generating and measuring shear waves in soft tissues. The advantage of using shear waves is that the shear wave velocity is proportional to the mechanical material properties of the tissues under investigation. Soft tissues are inherently viscoelastic and considerable effort has been made to quantify the dispersion, or variation of frequency, of the velocity and attenuation of shear waves. This is accomplished by measuring the shear wave motion and using Fourier-based techniques to extract the shear wave velocity and attenuation. Additional considerations are made in tissues where geometric dispersion is also present such as in the heart. Viscoelastic characterization of the shear viscoelasticity of soft tissues *in vivo* such as human liver, human kidney, and swine heart will be shown. We will also demonstrate parameterization of the results by using a model-free approach or by fitting the shear wave velocity dispersion to rheological models. The diagnostic value of the viscoelastic parameters will be discussed for each particular application. [This work was supported in part by NIH Grant Nos. DK092255, DK082408, and EB002167.]

Contributed Papers

8:40

2aPA3. Nonlinear surface acoustic waves on lithium niobate in microfluidic devices. Yurii A. Ilinskii, Evgenia A. Zabolotskaya, and Mark F. Hamilton (Appl. Res. Labs., Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

Surface acoustic waves (SAW) are used frequently in microfluidic devices. Normally SAWs are generated on the surface of a piezoelectric material. Commonly used PZT is not appropriate for biomedical applications because of its high lead content, over 60% by weight. In this talk, a study of nonlinear SAW propagation in a piezoelectric substrate is presented. Model equations describing nonlinear SAW propagation in a piezoelectric crystal are derived from first principles. Elastic, piezoelectric, dielectric, and electrostrictive properties of a crystal with arbitrary symmetry are taken into account. The derived evolution equations are integrated numerically to illustrate nonlinear distortion of an initially sinusoidal wave of finite amplitude. As an example, SAW propagation along the X axis on single crystal 127.680 YX-cut lithium niobate ($LiNbO_3$), referred to as 128-YX-LN, is considered. This $LiNbO_3$ cut is typically used in microfluidic devices because it provides large mechanical displacements in the substrate. Analysis of the nonlinearity matrix permits quantification of the relative contributions to surface wave distortion from each physical phenomenon. [Work supported by the IR&D program at ARL:UT.]

8:55

2aPA4. Shear wave propagation in worm-like micellar fluids. Josh R. Gladden, Rachel Crim, Amanda Gamble, and Cecille Labuda (Phys. & NCPA, Univ. of MS, 108 Lewis Hall, University, MS 38677, jgladden@olemiss.edu)

In viscous Newtonian fluids, support of shear waves are limited to the viscous boundary layer. Non-Newtonian fluids which have shear modulus, however, support shear waves over much longer distances. The restoring force responsible for the shear wave propagation arises from the entanglement of high aspect ratio macromolecules. We report low frequency (30–60 Hz) shear wave studies of aqueous worm-like micellar fluids composed of cetyltrimethylammonium bromide (CTAB) for the surfactant and sodium salicylate (NaSAL) as the salt over a wide concentration range (20–500 mM CTAB). Shear speeds range from 75 to 700 mm/s over this concentration range at room temperature with evidence of two phase transitions at 200 mM and 375 mM CTAB. Shear stress attenuation and temperature resolved measurements between 20 and 40 C will also be presented.

9:10

2aPA5. Using resonant ultrasound spectroscopy on samples with and without conducting coatings to measure piezoelectric constants. Rhiannon E. Vieceli and Julian D. Maynard (Phys., Penn State Univ., 104 Davey Lab., University Park, PA 16802, rev5028@gmail.com)

Piezoelectrics are often used at low temperatures, but among the large number of piezoelectric materials, only one (quartz) has had its properties measured at low temperatures. Because measuring piezoelectric constants with the traditional electrical impedance method has shortcomings, particularly at liquid helium temperatures, it would be advantageous to make measurements with resonant ultrasound spectroscopy (RUS). EerNisse and Holland (1967) established a theoretical basis and Ogi *et al.* (2002) demonstrated an experimental RUS method. A problem with RUS for piezoelectrics is that resonance frequencies are much more sensitive to elastic behavior than to piezoelectric behavior, so that extraordinary precision is required. However, one may make a RUS measurement on a sample twice, once with an electrically conducting coating on sample faces and once without, and analyze the differences in the frequency spectra. Because the effect of the conducting coating depends more on the piezoelectric behavior than on the elastic behavior, analyzing the frequency differences suppresses the dependence on the elastic constants and enhances the measurement of the piezoelectric constants. This paper will present theoretical and experimental results for this RUS method.

9:25

2aPA6. Measurement of dispersion and attenuation in granular media using a filter-correlation method. Caleb O'Connor and Andi Petculescu (Dept. of Phys., Univ. of Louisiana at Lafayette, Lafayette, LA, cso5597@louisiana.edu)

A wideband technique for measuring sound dispersion and frequency-dependent attenuation in granular media is presented. The measurements were done on a mono-disperse medium of 2-cm solid polypropylene balls, over the frequency range of 500 Hz–20 kHz, enough to cover both weak- and strong-scattering regimes. A horn driver was used to launch sound into a foam-lined bucket containing the granular medium. The latter was mechanically isolated from the driver so as to minimize direct-contact coupling. The foam isolation was not enough, especially at resonances of the bucket-granular system. To account for the mass loading of the bucket by the granulars, the response of the bucket wall was measured by laser Doppler vibrometry both without and with the granulars. The response of the granular medium itself was extracted from the overall response through successive measurements of the individual responses of the driver, driver + bucket, and driver + bucket + granular. The frequency-dependent wavenumber of the granular is obtained by a filter-correlation method, using the driver response as reference. After successive bandpass filtering, the phase speed and attenuation are obtained within each band, respectively, by signal alignment and amplitude log ratio.

9:40

2aPA7. Design of piezoelectric energy harvesting system using cantilever beam. Jin-Su Kim, Un-Chang Jeong, Sun-Hoon Lee, Jung-Min Jeong, and Jae-Eung Oh (Mech. Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, fermatajin@hanyang.ac.kr)

In this paper, a design for an energy harvesting device using cantilever beam will be investigated and experimental results will be presented to validate the design. The energy harvesting device in the study is 31-unimorph piezoelectric which was used to convert small amplitude mechanical vibration from a specific machine application into an electrical energy source that could be used for electronic devices with low power requirements. The primary purpose of the design is to illustrate a method to design a cantilever beam that is optimized for attached position of piezoelectric by Experiment and FEM. From the given vibration data a range of frequencies where the energy harvesting device will generate the greatest amount of energy is determined. The device is then designed specifically targeting that frequency range with sinusoidal wave about resonant frequency. And results of this study show the change trend of output voltage according to changing circuit elements. This approach is presented as part of a more general approach to designing energy harvesters for any application. Also, it will be shown how attached position of piezoelectric used for cantilever beam were chosen.

9:55

2aPA8. *In-situ* ultrasonic evaluation of structural/nuclear materials. K. Sakthipandi (Phys., Sethu Inst. of Technol., Pullor, Kariapatti, Tamil Nadu 626115, India, sakthipandi@gmail.com) and V. Rajendran (Ctr. for Nano Sci. and Technol., K S Rangasamy College of Technol., Namakkal, Tamil Nadu, India)

Physical properties of components through ultrasonic non-destructive evaluation play a vital role to understand in quality and strength materials and also help to extend life of the components. Measurements of ultrasonic velocity and attenuation as a function of temperature were used to reveal the structural/phase transitions, initiation and growth of fatigue-induced damages, and life-limiting fatigue crack during the aging of materials. Indigenously designed experimental set-ups was designed for *in-situ* ultrasonic velocities and attenuation measurement over a wide range of temperature from 120 to 300 K and 300 to 1200 K. The ultrasonic velocity/attenuation measurements carried out on AISI316 stainless steel, β-quenched zircaloy-2 specimen and maraging steel were used to explore the formation and resolution/recrystallization of intermetallic and coherent precipitations. Further, the ultrasonic velocity/attenuation measurements carried out in bulk and nano perovskites samples ($La_{1x}Sr_xMnO_3$, $Nd_{1x}Sr_xMnO_3$, $Sm_{1x}Sr_xMnO_3$ and $Pr_{1x}Sr_xMnO_3$) were used to explore the phase transition (TC), charge ordering (TCO), and Jahn-Teller (TJT) temperature. The bulk and nanocrystalline nature of the perovskites were explained based on observed anomalies at transition temperature. The plot of first derivative of temperature dependent ultrasonic parameters was used to reveal the precise information to detect the early stages of microstructural and substructure variations in material.

TUESDAY MORNING, 6 MAY 2014 BALLROOM B, 8:30 A.M. TO 11:25 A.M.

Session 2aPP

Psychological and Physiological Acoustics: Temporal Processing, Compression, and Cochlear Implants: Session in Honor of Sid P. Bacon

Neal F. Viemeister, Cochair
Psychology, Univ. of Minnesota, 75 E. River Pkwy, Minneapolis, MN 55455

Walt Jesteadt, Cochair
Boys Town National Res. Hospital, 444 N. 30th St., Omaha, NE 68131

Chair's Introduction—8:30

Invited Papers

8:35

2aPP1. Modulation masking within and across carriers for subjects with normal and impaired hearing. Brian C. Moore and Thomas Baer (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

Sid Bacon was a pioneer in studies of the extent to which the detection of target amplitude modulation (AM) of a carrier is affected by additional (masker) amplitude modulation applied to the same carrier (within-channel modulation masking) or to a different carrier (across-channel modulation masking). Here, these two types of modulation masking were compared for normal-hearing and hearing-impaired subjects. The target was either 4-Hz or 16-Hz sinusoidal AM of a 4000-Hz carrier. The target AM depth was fixed. The masker AM was applied either to the same carrier or to a carrier at 3179 or 2518 Hz. The masker AM rate was 0.25, 0.5, 1, 2, or 4 times the target rate. The masker AM depth was varied adaptively to determine the value needed just to mask the target AM. Preliminary results indicate that within-channel modulation-masking patterns are similar for normal-hearing and hearing-impaired subjects, suggesting that the hypothetical modulation filters are not affected by hearing loss. However, the amount of across-channel modulation masking is lower for normal-hearing than for hearing-impaired subjects, presumably because of the reduced frequency selectivity of the latter. The increased across-channel masking for the hearing-impaired subjects may contribute to their difficulties in understanding speech in background sounds.

9:00

2aPP2. Possible contribution of cochlear compression to amplitude modulation detection. Jungmee Lee (Dept. of Commun. Sci. and Disord., Univ. Wisconsin, 475 Goodnight Hall, 1975 Willow Dr., Madison, WI 53706, jmlee6@msn.com)

Among his many research areas, Sid Bacon's work on auditory temporal processing, and his effort to connect psychophysical phenomena with cochlear compression, contributed greatly to our understanding of auditory system function. Inspired by his work, I will present research suggesting that (1) temporal processing, as measured by amplitude-modulation (AM) detection, is better for people with cochlear hearing impairment (HI) than those with normal hearing (NH) when the age of groups and loudness of the stimuli is matched, (2) AM detection of a target is poorer when an AM masker is presented at frequencies in a region with hearing loss than in a region of normal hearing, (3) masking of AM detection by an AM masker is reduced in HI when the masker is compressed using a low-distortion compressor, but is unaffected for NH, and (4) amplitude modulation of 2f1-f2, Distortion Product Otoacoustic Emissions recorded with amplitude-modulated f1 and steady-state f2, is correlated with AM perception. Taken together, the results provide evidence that cochlear compression plays an important role in auditory temporal processing as measured by AM perception.

9:25

2aPP3. The role of pitch strength in extracting speech from complex backgrounds. Marjorie R. Leek (Res. Dept., VA Loma Linda Healthcare System, 11201 Benton St., Loma Linda, CA 92357, Leekmar@aol.com)

How do people extract a target speech signal from a chorus of other sounds including other speech sounds? And what are the central and peripheral auditory processes that make this possible, and that may fail in people with damaged auditory systems? These critical issues in speech and hearing science were of continuing interest to Sid Bacon throughout his career. As Sid has noted, the problem is multifaceted, and he chose to study many of the individual factors, as well as their interactions. In keeping with that approach, I will discuss how normal-hearing and hearing-impaired listeners use pitch to extract a target from a complex noise background. In one study, auditory stream segregation of iterated rippled noises (IRN) with varying pitch strengths was explored to understand the limits of tonality for separating two patterns of sounds. In a second study, voice pitch strength was investigated as a means to support perceptual separation of target and background speech, with a focus on either spectral or temporal characteristics of the speech sounds. Interactions between degree of tonality in speech and other factors related to perception of speech in background sound will be examined. [Work supported by NIH.]

9:50–10:05 Break

Chair's Introduction—10:05

10:10

2aPP4. Role of temporal fine structure in speech recognition: From psychoacoustics to cochlear implants. Frederic Apoux and Eric W. Healy (Speech & Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210, fred.apoux@gmail.com)

The path from fundamental research to real-world application is often long and sinuous. This presentation will depict how psychoacoustic studies performed in Sid Bacon's laboratory have engendered our original view of the role of temporal fine structure (TFS) in speech recognition, and how this view has in turn led to the development of an extremely effective speech-processing strategy for cochlear implants. First, psychoacoustic data showing how normal-hearing listeners can detect amplitude modulation when presented with only the TFS will be described. Second, a series of experiments illustrating the "absence" of acoustic speech cues in the TFS will be briefly presented. Finally, the results of a recent study involving a dual-carrier vocoder will be described. These results suggest that TFS cues are primarily used to assist in identifying which auditory channels are dominated by the target signal so that the output of these channels can be combined at a later stage to reconstruct the internal representation of that target. They also indicate that cochlear implants implementing a speech-processing strategy based on the "dual-carrier strategy" have the potential to restore nearly perfect speech intelligibility in noise. [Work supported by NIH.]

10:35

2aPP5. Understanding the benefits of electric-acoustic stimulation. Christopher Brown (Dept. of Commun. Sci. and Disord., Univ. of Pittsburgh, 4033 Forbes Tower, 3600 Forbes at Atwood, Pittsburgh, PA 15260, cbrown1@pitt.edu)

The problem of speech understanding in the presence of background noise is a difficult one, especially for users of cochlear implants. Although these users can often perform well on speech tasks in quiet, they typically show rapid declines in background noise. The retention of low-frequency residual acoustic hearing has been shown to provide significant benefit in this regard. Data will be presented on our work on this topic, from characterizing the benefit to exploring ways of providing it to cochlear implant users who do not show a benefit typically.

11:00

2aPP6. Spectral resolution and its effects on temporal analysis in cochlear-implant perception. Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Sid Bacon's contributions to auditory science span a wide range of topics but two areas were constant themes throughout his career: temporal modulation perception, and peripheral processing and frequency selectivity. In this study, speech understanding in noise was measured, with a focus on the role of inherent temporal fluctuations in noise maskers. Tonal maskers, presented to cochlear-implant users, were placed at the center frequencies of each frequency channel of the implant, thereby producing the same masker energy as a noise masker in each frequency channel, but without the inherent fluctuations. In contrast to the results from normal-hearing subjects

listening through a tone-excited envelope vocoder, cochlear-implant users gained no benefit from eliminating the inherent fluctuations from the maskers. Further experiments suggested that the poor spectral resolution of cochlear implants resulted in a smoothing of the temporal envelope of the noise maskers. The results indicate an important, and potentially overlooked, effect of spectral resolution on the temporal representations of speech and noise in cochlear implants. The results also suggest a new interpretation for why cochlear-implant users, and perhaps hearing-impaired listeners, generally show reduced masking release when additional temporal modulations are imposed on noise maskers. [Work supported by NIH grant R01DC012262.]

TUESDAY MORNING, 6 MAY 2014 — 553 A/B, 8:20 A.M. TO 11:30 A.M.

Session 2aSA

Structural Acoustics and Vibration, Physical Acoustics, Engineering Acoustics, and Noise: Acoustic Metamaterials I

Christina J. Naify, Cochair
Acoust., Naval Res. Lab., 4555 Overlook Ave. SW, Bldg. 2, 138G, Washington, DC 20375

Michael R. Haberman, Cochair
Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Invited Papers

8:20

2aSA1. Metal based acoustic metamaterials. Andrew Norris, Adam J. Nagy, and Alexey S. Titovich (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu)

Two apparently distinct types of acoustic metamaterial are considered: a metallic phononic lattice structure and an array of metal shells in water. The unifying feature is that metal acts primarily as a stiffness, and by adding material one arrives at a desired effective density. The metal provides a reservoir of stiffness in the sense that a little bit goes a long way toward the effective stiffness of water, or properties close to water. We first describe the Metal Water structure proposed as a generic metamaterial for transformation acoustics, in both 2D and 3D. The structures have isotropic elastic properties with low shear modulus, hence mimicking water. While designed for long-wavelength effective properties the structures also display interesting finite frequency effects, such as negative index properties. The thin shell metamaterial elements achieve the bulk modulus of water at a specific thickness/radius ratio. Simultaneous matching of effective bulk modulus and density is obtained using an internal mass. By design, both types of metamaterials separate stiffness and density, allowing for simple lumped parameters modeling. The use of metal also has implications for optimal cloaking properties, which result from the fact the metallic structure is non-causal. [Work supported by ONR.]

8:40

2aSA2. Transparent acoustic metamaterials for broadband aqueous applications. Theodore P. Martin (Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, theodore.martin@nrl.navy.mil), Christina J. Naify, Christopher N. Layman, Charles A. Rohde (National Res. Council, Washington, DC), Abel Thangawng, Michael Nicholas, David C. Calvo, and Gregory J. Orris (Naval Res. Lab., Washington, DC)

Matched impedance between different components is an integral part of many wave transport device applications. Achieving transparency in acoustics has typically been accomplished using bulk materials such as ρc rubbers or resonance phenomena over narrow bandwidths. This talk will focus on broadband acoustic metamaterials that are impedance matched to water and that offer a broad range of locally definable sound speeds. Experimental investigations of a number of device applications will be presented, including a transparent gradient index (GRIN) lens, an omnidirectional focusing coating, and elastic lattices that mimic the material properties of water. Impedance-matching over a wide range of sound speeds is achieved by changing the filling fraction of sub-wavelength acoustic scattering components that are individually impedance-matched to water. Both sonic crystals and pentamode trusses are explored as component lattices to produce fluid-like transport over a broad bandwidth in the homogenization limit of the lattices. Excellent agreement is obtained between predictions using fully elastic homogenization theory and the acoustic multistatic signatures measured in the vicinity of the as-realized devices. [Work supported by the Office of Naval Research.]

9:00

2aSA3. From acoustic metamaterials to functional metasurfaces. Nicholas X. Fang, Jun Xu, Chu Ma, and Navid Nemati (MechE, MIT, 77 Massachusetts Ave., Cambridge, MA 02139, nicfang@mit.edu)

By guiding and controlling the wave path through a deformed space, metamaterial devices that display distinctive response to light, acoustic waves, and heat waves have opened up a new field of considerable interest. However, current challenge issues such as high loss and complex geometries are hampering the development of metamaterial technology. Is it possible to tailor the field information

contained by a complex volumetric object with a thin metamaterial structure, so the observer from afar would not tell the difference? In this invited talk, we will present our research progress toward tailoring the edge rays and creeping rays with acoustic metasurfaces. The tailored metasurface could lead to two pronounced effects: redirecting the reflected rays by spatial impedance gradient and reducing the strength of the edge rays. In fact our recent study suggest such illusional effects by embedding wedges in a transformed acoustic medium, and we will present theoretical analysis and experimental study of such engineered components with acoustic metasurface. The potential application of such novel device concept in underwater communication and medical ultrasound will be also discussed.

9:20

2aSA4. Pressure-invariant non-reflective and highly dissipative acoustic metamaterials. Alireza Amirkhizi (Mech. Eng., Univ. of Massachusetts, Lowell, Perry Hall 331, 197 Riverside St., Lowell, MA 01854, alireza_amirkhizi@uml.edu), Christian Nielsen, Zhanzhan Jia, Wiroj Nantasetphong, Hossein Sadeghi, Kristin Holzworth (Mech. and Aerosp. Eng., Univ. of California, San Diego, La Jolla, CA), Ankit Srivastava (Mech., Mater., and Aerosp. Eng., Illinois Inst. of Technol., Chicago, IL), and Sia Nemat-Nasser (Mech. and Aerosp. Eng., Univ. of California, San Diego, La Jolla, CA)

Metamaterials have shown great potential to transform the design of acoustic components in many applications. Many composites with extreme properties have been envisioned, designed, fabricated, and experimentally verified. The next step involves testing such composites in realistic application environments. Oceans are one such environment in which the mechanical effects of water pressure and flow become important factors in any acoustic design, particularly for soft dissipative shells. We have designed a layered metamaterial composite that not only shows very high dissipation but also matches the acoustic impedance of water. Furthermore, we have experimentally verified that the relevant properties of the constituents of this layered design do not change under pressure levels that exist down to significant depths. We are in the process of fabricating this composite to test its acoustic properties under pressure. The metamaterial composites lend themselves naturally to multi-component designs, examples of which as well as gradient media will be presented. Some potential novel applications of gradient and layered components will be discussed.

9:40

2aSA5. Wave propagation in three dimensional crystalline foams. Alessandro Spadoni (Mech. Eng., EPFL, STI-IGM-LOMI, Station 9, Lausanne, 1015 Lausanne, Switzerland, alex.spadoni@epfl.ch)

Recent progress in manufacturing of crystalline foams has introduced cellular solids with very low relative density, the portion of volume occupied by the solid phase. For closed-cell configurations, this means thin films enclosing entrained fluid. While numerous mechanical models for the mechanical properties of 2D, open-cell configurations have been proposed, 3D closed-cell configurations are described by phenomenological models based on powers of the relative density due to their complexity. Elastic wave propagation in such media presents similar challenges and is described by Biot's theory, a model derived from a strain-energy functional defined at the macroscale, based on averaged microstructural quantities. Biot's theory requires equivalent mechanical properties for drained and undrained configurations which are often not available. Exploiting periodicity, we developed a detailed finite-element model to explicitly describe the coupling of fluid and solid. The entrained fluid is compressible, inviscid, and both convection and heat conduction are neglected. Three crystalline foams are considered: Kelvin, rhombic, and Weaire-Phelan configurations. In this talk, I will discuss frequency regimes with a single and two longitudinal pressure wavemodes, and super anisotropy. Dispersion depends on two key frequencies: film resonant frequency, and the natural frequency of a pore with deformable walls.

10:00–10:30 Break

Contributed Papers

10:30

2aSA6. Tapered labyrinthine acoustic metamaterials for coherent controlling of acoustic wave. Yangbo Xie, Adam Konneker, Bogdan-Ioan Popa, and Steven A. Cummer (Duke Univ., 3417 CIEMAS, Durham, NC 27705, yx35@duke.edu)

Acoustic metamaterials with their exotic material properties enable unprecedented control over acoustic wave propagation and reflection. Besides utilizing locally resonating structures or non-resonant composite effective media, non-locally resonating spatial coiling structures have recently been adopted to design negative or high positive refractive index metamaterials. We have in the past experimentally demonstrated the unit cell characteristics of one kind of labyrinthine metamterial (Xie *et al.* PRL 2013) and its impedance matching improved versions (Xie *et al.* APL 2013). In this work, we present several coherent modulation devices based on our recently proposed tapered labyrinthine metamaterials. With thickness of only one or two metamaterial cell layers, we can create an acoustic blazed diffraction grating, a phase conjugation lens, or a flat lens that can perform plane wave-cylindrical wave conversion. The design process and experimental demonstrations will be presented. The coherent controlling devices feature precise phase modulation, high-energy throughput, broad operating bandwidth, and sub-wavelength thickness. Our work demonstrates that labyrinthine metamaterials can be the unit cells of choice for functional coherent acoustic modulation devices.

10:45

2aSA7. Ultrasonic subwavelength focusing above periodic membrane arrays in immersion. Shane Lani, Karim G. Sabra, and F. Levent Degertekin (Woodruff School of Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr., Atlanta, GA 30332, shane.w.lani@gmail.com)

Subwavelength focusing and imaging has been a long sought after goal and one that metamaterials can possibly achieve. In 2011, Lemoult *et al.* used time reversal techniques to focus sound to as small as $\lambda/25$ in air by using the evanescent wave field above a gird of soda cans acting as Helmholtz resonators [Lemoult *et al.* Phys. Rev. Lett. **107**, 064301, (2011)]. This paper will demonstrate subwavelength focusing in immersion in the 11–0 MHz frequency range with capacitive micromachined ultrasonic transducer (CMUT) arrays. CMUTs are microscale (10–100 μm wide) membrane arrays, which support evanescent surface waves that derive their dispersive properties not only from the periodic structure of the array, but also from the membrane resonance. Furthermore, CMUTs have embedded electrodes for electrostatic excitation and detection of acoustic waves which allow implementation of time reversal techniques to focus the dispersive evanescent surface waves using only the CMUTs on the same substrate as sources and receivers. Using a finite boundary element method simulation, we demonstrate subwavelength focusing at points in the near-field above a 2D CMUT array in immersion.

11:00

2aSA8. Acoustic metamaterial elements from tunable elastic shells. Alexey Titovich and Andrew Norris (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, alexey17@eden.rutgers.edu)

Elastic shells are used as elements in novel acoustic metamaterials. Tuning the shells produces unnatural and favorable acoustic qualities in the quasi-static regime. This is achieved by internally stiffening the shell with an axisymmetric distribution of springs which connect the added central mass to the shell. The two parameters: stiffness and mass are carefully optimized for the desired effect. Flexural resonances of the shell dominate the frequency response, but are constrained in the quasi-static regime as shown by the analytical model. As an example of transparency, an aluminum shell of radius 1 cm is tuned to water with and acrylic internal oscillator exhibiting a near-zero scattering cross section up to ka = 0.6. Also, individually tuning each shell in a fluid saturated array is a means of creating devices based on transformation acoustics such as a cylindrical to plane wave lens. Investigations of favorable high frequency effects and active tuning are presented. Another method of changing the effective acoustic properties of a shell is to attach a second shell to the inside creating a composite structure. The thickness of each shell is optimized to yield desired effective medium properties. Tuning to water yields a broad frequency range of transparency.

11:15

2aSA9. Physical constraints on lossy acoustic metamaterials with complex effective properties. Caleb F. Sieck, Michael R. Haberman (Appl. Res. Labs., The Univ. of Texas at Austin, 4021 Steck Ave. #115, Austin, TX 78759, cfsieck@utexas.edu), and Andrea Alù (Dept. of Elec. & Comput. Eng., The Univ. of Texas at Austin, Austin, TX)

Recent theoretical and experimental work on acoustic metamaterials (AMM) has demonstrated materials that demonstrate many properties, such as negative modulus and density, beyond what is capable using conventional materials. In most cases, AMM are assumed to be passive and causal with frequency dependent losses accounted for via complex modulus and density. Despite the maturity of AMM research, literature concerning the physical constraints on the complex effective constitutive properties for passive, causal AMM is very limited. This work presents the physical limits for the real and imaginary effective dynamic mass density and dynamic compressibility via recourse to restrictions placed on the AMM by conservation of energy, passivity, and causality. We further note that constitutive properties are determined from the effective wavenumber and impedance extracted from simulation or experiment. Although care is normally taken to guarantee that passivity holds for the wavenumber and impedance, assumptions implicit in various homogenization schemes can result in constitutive properties that do not satisfy passivity and causality. This work will therefore also discuss implications on AMM homogenization and extraction of properties due to constraints based on the foundational concepts of conservation of energy, passivity, and causality. [This work was supported by the Office of Naval Research.]

TUESDAY MORNING, 6 MAY 2014 BALLROOM A, 8:00 A.M. TO 12:00 NOON

Session 2aSC

Speech Communication: Speech Perception I (Poster Session)

Sayako Earle, Chair

Dept. of Speech, Lang. and Hearing Sci., Univ. of Connecticut, 123 Davis Rd., Storrs, CT 06268

All posters will be on display from 8:00 a.m. to 12:00 noon, To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10.00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

2aSC1. Gradient perception of within-category nasality in English vowels. Georgia Zellou and Delphine Dahan (Linguist, Univ. of Pennsylvania, 800 N. 48th St., #26, Philadelphia, PA 19139, gzellou@sas.upenn.edu)

Nasality encodes a phonemic consonant contrast in English, and its acoustic correlates affect adjacent vowels. The phonological status of nasality predicts that people encode vowel nasality as a discrete and binary feature, i.e., as the presence or absence of velum lowering. The present study examined whether listeners can also infer the degree of velum lowering encoded in a vowel. We resynthesized 20 word pairs that contrasted in vowel nasality only (e.g., bet vs. bent [produced with no nasal murmur]) and created, for each pair, a continuum of seven stimuli varying linearly in their degree of vowel nasality. The 140 stimuli were used to assess listeners' categorization and discrimination. First, participants categorized each stimulus as oral or nasal. They were then presented with two stimuli selected from the same continuum and judged whether they consisted of physically identical or different tokens. Using linear-regression modeling, we demonstrated that discriminating between two stimuli is determined by their acoustic distance in velum height above and beyond their category assignment. Thus, people can encode velum height in stimuli they categorize identically. This finding adds to the growing body of evidence that listeners track properties of speech that go beyond establishing lexical contrasts.

2aSC2. Arrays of subcritical width rectangular speech bands maintain intelligibility at high intensities. Richard Warren, James Bashford, and Peter Lenz (Psych., Univ. of Wisconsin-Milwaukee, PO Box 413, Milwaukee, WI 53201, rmwarren@uwm.edu)

Speech intelligibility declines at high intensities for both normally hearing and hearing impaired listeners. However, it appears that this rollover can be minimized and intelligibility preserved by reducing speech in high frequency regions to an array of noncontiguous bands having vertical filter slopes (i.e., rectangular bands) and widths substantially narrower than a critical band. Normally hearing listeners were presented with sentences consisting of a 500-Hz lowpass pedestal band and an array of ten 4% bands spaced at 1/3-octave intervals from 1000 Hz to 8000 Hz. The pedestal band was fixed at 70 dB and the subcritical-band array varied from 55 to 105 dB in peak level. Desired sub-ceiling intelligibility ranged from 80 to 89% and was statistically asymptotic for levels from 75 to 105 dB. The largest intelligibility difference across that range, found between 85 and 105 dB, was just 1.6%. For that same contrast in levels, Molis and Summers [ARLO **4**, 124–128 (2003)] obtained a significant intelligibility loss of 26.7% for spectrally continuous highpass speech. It is suggested that subcritical-width bandpass filtering reduces rollover by limiting firing rate saturation to a subset of fibers comprising individual critical bands. Implications for hearing aid construction will be discussed. [Work supported by NIH.]

2aSC3. Asymmetries in vowel perception: Do they arise from focalization, perceptual magnets, or both? Matthew Masapollo and Linda Polka (McGill Univ., 1266 Pine Ave. West, Montreal, QC H3G 1A8, Canada, matthew.masapollo@mail.mcgill.ca)

While directional asymmetries are ubiquitous in cross-language studies of vowel perception, their underlying mechanisms have not been established. One hypothesis is that listeners display a universal perceptual bias favoring vowels with greater formant frequency convergence, or focalization (Polka and Bohn, 2011). A second, but not mutually exclusive, hypothesis is that listeners are biased toward prototypical vowel exemplars in their native language (Kuhl, 1993). In a test of these hypotheses, English listeners discriminated synthesized English /u/ and French /u/ vowels presented in pairs. While the French /u/ tokens exhibit greater formant convergence (between F1 and F2), English listeners have previously been shown to rate the English /u/ tokens as "better" instances of the category (Molnar, 2010). Preliminary results demonstrate that the degree of focalization affects vowel discrimination. When discriminating vowel changes presented in the direction going from the more focal (French) to less focal (English) /u/ vowels, English listeners' reaction times were slower, relative to the same changes presented in the reverse direction. These results suggest that listeners treat the more focal vowels as perceptual reference points. Additional data collection with French listeners is ongoing. The implications of these findings for theories of vowel perception will be discussed.

2aSC4. Stream segregation of concurrent speech and the verbal transformation effect: Influence of fundamental frequency and lateralization cues. Marcin Stachurski, Robert J. Summers, and Brian Roberts (Psych., School of Life & Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, r.j.summers@aston.ac.uk)

Listening to a repeating recorded word produces verbal transformations (VTs); perceptual re-grouping of phonetic segments may contribute to this effect. The influence of fundamental frequency (F0) and lateralization grouping cues was explored by presenting two concurrent sequences of the same word resynthesized on different F0s (100 and 178 Hz). In experiment 1, listeners monitored both sequences simultaneously, reporting for each any change in stimulus identity. Three lateralization conditions were used—diotic, interaural time difference ($\pm$680-μs ITD), and dichotic. The results were similar for the first two, but fewer forms and later transformations were reported in the dichotic condition. This suggests that large lateralization differences between the two sequences per se have little effect—rather, there are more possibilities for perceptual re-grouping when each ear receives both sequences. For all conditions, VTs reported on one sequence were mainly independent of the other. Experiment 2 investigated the effect of number of sequences presented and monitored. The most forms and earliest transformations were reported when two sequences were presented but only one was monitored, indicating that high task demands reduce reporting of VTs for concurrent sequences. Overall, these findings support the idea that perceptual re-grouping contributes to the VT effect. [Work supported by EPSRC.]

2aSC5. Does a perceived intensity cause pitch change in noise-vocoded vowels? Marina Takabayashi (Sensory and Cognit. Neural System Lab., Faculty of Life and Medical Sci., Doshisha Univ., 15-11 Okenoi-cho,Takeda, Fushimi-ku, Kyoto-shi 612-8421, Japan, bmk1086.splash@gmail.com), Kohta I. Kobayasi, and Hiroshi Riquimaroux (Sensory and Cognit. Neural System Lab., Faculty of Life and Medical Sci., Doshisha Univ., Kyotanabe, Japan)

Noise-vocoded speech sound (NVSS) is the synthesized speech sound whose frequency information is greatly reduced while the amplitude envelope information remains preserved. In NVSS the fundamental frequency does not exist, and a change in amplitude is perceived as a change in not only loudness but also pitch. Original speech having physically same amplitude is not always identical. The phenomenon might occur also in NVSS. The purpose of this study is to examine whether change in not amplitude but loudness creates pitch change in NVSS. Subjects listened to paired original speech Japanese vowels and judged whether the second vowels were perceived louder or softer than the first ones. And they listened to paired noise-vocoded vowels to evaluate whether pitch of the second sounds rises or falls from the first ones. Results show that changes in loudness seem to cause pitch change in NVSS. After this, loudness perception in NVSS will be investigated. The data will show whether pitch of the second sounds change from the first ones or not when loudness of the first and the second sounds are same. And loudness perception in noise-vocoded vowels will be compared with loudness perception in the original speech.

2aSC6. Perceived emotional valence in clear and conversational speech. 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)

In the laboratory, talkers asked to speak as though talking to an individual with hearing loss modify their speech from their everyday conversational style to a "clear" speaking style. In the real world, individuals with hearing loss sometimes complain that their frequent communication partners seem to be shouting at them, while the communication partners insist that they are just trying to speak more clearly. Acoustic analyses have contrasted angry speech with neutral speech and clear speech with conversational speech. A comparison of these analyses reveals that angry speech and clear speech share several acoustic modifications. For example, both clear speech and angry speech show increased energy at high frequencies. The present study will explore whether clear speech sounds angry to listeners. Young adult listeners with normal hearing 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 no emotion). The resulting data will show whether clear speech is more likely to be judged as sounding angry than typical conversational speech.

2aSC7. Error analysis and modifications to the short-time speech transmission index. Karen Payton and Matthew Ferreira (Elec. & Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747-2300, kpayton@umassd.edu)

The Speech Transmission Index (STI) predicts the intelligibility of speech degraded by noise and reverberation. Recently, Payton and Shrestha [J. Acoust. Soc. Am. **134**, 3818–3827 (2013)] reported on a short-time speech-based STI (ssSTI) to predict time-varying intelligibility of speech using analysis windows shorter than 1 s. While the ssSTI generally tracked a theoretical STI calculated using octave-band signal-to-noise ratios, it deviated from the theoretical calculation for windows shorter than 0.3 s. The current work analyzes and improves the performance of the ssSTI for speech degraded by stationary speech-shaped noise. Using a cluster analysis, the time-varying standard deviation was determined to be inversely proportional to window length, octave band and speech envelope variance. Two ssSTI modifications are proposed to reduce the 0.3 s window limitation: A silence detection algorithm eliminates non-zero ssSTI values that occur during silence and a modified envelope extraction scheme reduces the standard deviation by increasing envelope bandwidth. Using the 0.3 s window as a performance benchmark, new octave-band specific window limitations, ranging from 151 ms to 21 ms, were established. The modified ssSTI also works with common octave-band window lengths as short as 30 ms when full envelope bandwidths are used in combination with the silence detector.

2aSC8. Using the short-time speech transmission index to predict speech reception thresholds in fluctuating noise. Matthew Ferreira and Karen Payton (Elec. & Comput. Eng., Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., North Dartmouth, MA 02747, mjferreira128@gmail.com)

The Speech Transmission Index (STI) predicts the intelligibility of speech degraded by noise and reverberation. Recently, Payton and Shrestha [J. Acoust. Soc. Am. **134**, 3818–3827 (2013)] reported on the ability of a short-time speech-based STI (ssSTI) to predict the intelligibility of speech in the presence of fluctuating noise using analysis windows shorter than 1 s. They found the ssSTI highly correlated with theoretical STI calculations using windows as short as 0.3 s. In the current work, extended versions of the ssSTI were investigated for their ability to improve speech intelligibility prediction in the presence of fluctuating noise; a condition for which the long-term STI incorrectly predicts the same intelligibility as for stationary noise. No STI metric predicts a normal-hearing listener's improved ability to perceive speech in the presence of fluctuating noise as compared to stationary noise at the same signal-to-noise ratio. The investigated technique used window lengths that varied with octave band, based on human auditory temporal resolution as in the Extended Speech Intelligibility Index [Rhebergen and Versfeld, J. Acoust. Soc. Am. **117**, 2181–2192 (2005)]. An extended sSTI using speech-shaped noise instead of speech as a probe predicted published speech reception thresholds for a variety of conditions.

2aSC9. Gradient coding of voice onset time in posterior temporal cortex. Nathaniel D. Anderson (Beckman Inst., Univ. of Illinois, 1412 Beckman Inst., 405 N. Mathews Ave., Urbana, IL 61801, nandrsn3@illinois.edu), Joseph C. Toscano, Monica Fabiani, Gabriele Gratton, and Susan M. Garnsey (Beckman Inst., Univ. of Illinois, Urbana-Champaign, IL)

The issue of whether early stages of speech processing are influenced by category has been central to work in speech perception for decades. We present the results of an experiment using fast diffusive optical neuroimaging (Gratton and Fabiani, 2001, Int. J. Psychophysiol.) to address this question directly by measuring neural responses to speech with high temporo-spatial resolution. We found that changes in voice onset time (VOT) along a /b/-/p/ continuum evoked linear changes in neural responses in posterior superior temporal gyrus (pSTG) 100 ms after stimulus onset. This is the first non-invasive observation of such responses in humans. It is consistent with results from recent event-related potential (Toscano *et al*., 2010, Psychol. Sci.) and fMRI (Blumstein *et al*., 2005, J. Cognit. Neurosci.) studies, and provides evidence that those results reflect listeners' early encoding of speech sounds in pSTG, independently of phonological categories. Thus, the results provide evidence that speech perception is based on continuous cues rather than discrete categories. We discuss these results in light of recent intra-cranial EEG studies reporting either categorical effects in pSTG (Chang *et al*., 2010, Nature Neurosci.) or evidence that pSTG maintains fine-grained detail in the signal (Pasley *et al*., 2012, PLoS Biol.).

2aSC10. Effects of linguistic structure on perceptual attention given to different speech units. Shinae Kang and Keith Johnson (Linguist, UC Berkeley, 1203 Dwinelle Hall, UC Berkeley, Berkeley, CA 94720-2650, sakang2@berkeley.edu)

Listeners can shift their attention to different sizes of speech during speech perception. This study extends this claim and investigates if linguistic structure affects this attention. Since English has a larger syllable inventory than Korean and Japanese, each phoneme plays a larger functional role. Also, listeners have different levels of phonological awareness due to the differences in the orthography. We focus on the effect of perceptual attention on the perceptibility of intervocalic consonant clusters (VCCV) and whether it varies cross-linguistically by these structural factors. We first recorded eight talkers saying VC- and CV-syllables and spliced the syllables to create non-overlapping VC.CV-stimuli. Listeners in three language groups (English/Korean/Japanese) participated in a 9-Alternative-Forced-Choice perception task. They identified the CC as one of 9 alternatives ("pt", "pk", "pp", etc.) and in an attention-manipulated condition did the same task while also monitoring for target talkers. The preliminary result shows that Korean listeners showed less perceptual sensitivity to clusters than English listeners. Also, the English listeners showed better perception of syllable coda when prompted to focus on coda only. The result indicates that the linguistic structure of a language can potentially affect the level of perceptual attention that its users give to a linguistic unit.

2aSC11. Perception of dialectal prosody in Taiwan Mandarin. Mao-Hsu Chen (Linguist, Univ. of Pennsylvania, 4200 Spruce St., Apt. 310, Philadelphia, PA 19104, chenmao@sas.upenn.edu)

This pilot study aims at answering whether prosodic cues alone can account for the differences among three regional dialects in Taiwan Mandarin, Northern, Central, and Southern, which all belong to Mandarin Chinese. Assumed that prosodic cues alone can be used for distinguishing among different Taiwan Mandarin dialects, a perception experiment was conducted. The Northern dialect was best recognized while the identification rates of the Central and the Southern dialects were slightly below chance level. Results showed the tendency that it was easier for listener to identify his or her own dialect than it was to detect other regional dialects. Gender effect was observed to play a role in the recognition of the three dialects, which led to the following-up production experiment intended for exploring the acoustic differences among these three dialects. Preliminary results examined the descriptive tonal patterns of all four lexical tones in three Taiwan Mandarin dialects. The tonal registers produced by the Northern dialect speakers were more prominent than those of the Central and the Southern dialects in that they had the highest normalized F0 values for T1 tokens and greater pitch range, or steepest slope, for the other three contour tones, compatible with the result of the perception experiment.

2aSC12. Unpredictable and unintelligible: Individual differences in predictability-based reduction affect speech intelligibility. Rory Turnbull (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43215, turnbull@ling.osu.edu)

Words spoken in high predictability (HP) contexts tend to be phonetically reduced and less intelligible than words spoken in lower predictability (LP) contexts. However, individual differences in degree and strategies of reduction, and their effects on intelligibility, are largely unexplored. This study examined the role of autistic traits in speech intelligibility. Sixteen participants completed a speech in noise word identification task at 2dB SNR. Stimuli were recordings of words spoken in HP and LP contexts, extracted from sentences produced by talkers ranging in autism-spectrum quotient (AQ) scores. After the identification task, listeners also completed the AQ questionnaire. A higher AQ score indicates a greater prevalence of autistic traits in one's cognitive style. Logistic mixed effect regression modeling revealed an expected effect of duration, such that longer words were more likely to be correctly identified. Further, talkers with higher AQ scores were less intelligible than talkers with lower AQ scores. Unexpectedly, no effect of predictability condition was observed: LP words were not consistently more intelligible than HP words. No effect of individual differences in listener AQ scores was observed. These results suggest that predictability-based enhancement and reduction strategies vary between individuals and are not necessarily for the benefit of the listener.

2aSC13. Multidimensional scaling and order asymmetry of the acoustic change complex to voiceless fricatives in Polish, English, and Finnish listeners. Emma Brint, Paul Iverson (Speech, Hearing and Phonetic Sci., Univ. College London, 1 High Rd., London E182QN, United Kingdom, emma.brint.11@ucl.ac.uk), and Anita Wagner (Dept. of Otorhinolaryngology, Univ. Medical Ctr. Groningen , Groningen, Netherlands)

The acoustic change complex (ACC) is a P1-N1-P2 onset response to changes between sounds measured using EEG, which when measured at a central location, is not thought to be affected by the language experience of the listener. Recent research has shown that with a pair of stimuli, there is an asymmetry between the ACC recorded when they are presented in one order compared to the other. This study used eight voiceless fricatives to measure the ACC to all possible pairs from EEG recordings of 15 English, Finnish, and Polish listeners. The ACC magnitude was used to perform multidimensional scaling to create a two dimensional perceptual space for each language that is driven by the spectral characteristics of the stimuli.

Asymmetry occurred in all three languages depending on how peaky the first and second stimuli were. This asymmetry effect was greater for the Polish group, indicating that the Polish listeners' language experience increases their sensitivity to the spectral peakiness of fricatives. These results show that the ACC is in fact susceptible to native language effects, enhancing its application as a test for auditory speech processing.

2aSC14. The perception of postvocalic English stops in diphthongs and monophthongs using gating experiment. Siriporn Lerdpaisalwong (of Linguist, Univ. of Wisconsin-Milwaukee, 1915 E. Kenilworth Pl., Mailbox no. 83, Milwaukee, WI 53202, siriporn@uwm.edu)

There have been many studies on the acoustic cues in the preceding vowels for the perception of the final consonants (Hillenbrand *et al.*, 1983; Warren and Marslen-Wilson, 1988). Yet, none of the studies has been conducted to see whether there are any differences in listener perception of a final consonant in areas of diphthongs versus monophthongs. Since the average duration of diphthongs /eɪ/ and /oʊ/ is longer than that of monophthongs /i/ and /u/ (Hillenbrand *et al.*, 1995), this study investigates whether a listener perceives postvocalic English stops /p, t, k/ in monophthongs /i/ and /u/ faster than in diphthongs /eɪ/ and /oʊ/ using the gating paradigm. Fifteen American English speakers participated in this study (6F, 9M; 18–44 y.). The stimuli consist of 24 CVC words and noncewords with final /p, t, k/. Vowels of 24 words were chopped into ten gates each. The results show that the listeners perceived the final stops at about the same gates (areas) in both diphthongs and monophthongs in both word types. The results also show no significant difference among the perception of three stops, except between /p/ and /k/ in the real words with monophthongs.

2aSC15. Sonority of adjacent segments in the perception of durational distinctions. Olga Dmitrieva (Purdue Univ., 100 North University St., Beering Hall, Rm. 1289, West Lafayette, IN 47907, odmitrie@purdue.edu)

The distributional typology of length contrasts in consonants suggests that the sonority of adjacent segments may be relevant for the perception of duration differences in consonants. Across languages, short consonants typically contrast with long consonants intervocalically or next to high sonority consonants, such as glides or liquids. It has been proposed that surrounding sonorants facilitate the perception of durational distinctions by providing clear acoustic cues to the beginning and the end of the target consonant, which makes it easier to estimate the target's duration (Bradley, 2001; Padgett, 2003). The present study investigates this hypothesis by examining the effect of adjacent segments' sonority on the perception of durational differences in alveolar voiceless stops. Target stops of two lengths were placed in the environment of a preceding or following vowel, liquid, nasal, fricative, or non-homorganic stop and presented to listeners in a discrimination experiment. Initial results indicate an important role for syllable structure and its sonority profile in duration discrimination: Listeners appear to be more sensitive to length distinctions when stops targets are in the onset position preceded by a higher-sonority coda (e.g., al.ta) compared to stop targets in the coda position followed by a higher-sonority onset (e.g., at.la).

2aSC16. On the gradience in perceptibility of word-final voicing contrast in Russian. Mayuki Matsui (Linguist, Hiroshima Univ., 1-2-3 Kagamiyama, Higashi-Hiroshima-shi, Hiroshima 739-8522, Japan, matsui-ma@hiroshima-u.ac.jp)

Russian is one of the languages in which the underlying voiced obstruents devoice in word-final position, resulting in voicing neutralization (e.g., /rok/ [rok] "fate" vs. /rog/ [rok] "horn"). However, recent studies have shown that word-finally devoiced (i.e., underlying voiced) obstruent and the underlying voiceless one are acoustically different (Chen 1970, Dmitrieva *et al.* 2010, among others). That is, word-final devoicing shows a case of incomplete neutralization. Also, those acoustic differences are in some degree perceptible for listeners (Matsui 2011, Kharlamov 2012). This paper presents a perceptual analysis of incompletely neutralized obstruents in Russian. Pseudo-nouns produced by the native speakers of Russian were presented to the listeners as auditory stimuli. Sixteen native listeners identified what they heard in a forced-choice identification task. The most striking result to be reported in this paper is that the listeners' sensitivity to the underlying voiced and voiceless stimuli is different between obstruent types: the underlying voicing contrast in stops is harder to perceive than that in fricatives for listeners. Other than obstruent type, the effects of the stimuli presentation type and of the magnitude of the acoustic difference will also be discussed.

2aSC17. Discriminability and perceptual saliency of acoustic cues for final consonant voicing in simulations of cochlear-implant and electric-acoustic stimulation. Ying-Yee Kong (Dept. of Speech Lang. Pathol. & Audiol., Northeastern Univ., 226 Forsyth Bldg., 360 Huntington Ave., Boston, MA 02115, yykong@neu.edu), Ala Mullangi (Bioeng. Program, Northeastern Univ., Boston, MA), Matthew Winn (Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI), and Gail Donaldson (Dept. of Commun. Sci. and Disord., Univ. of South Florida, Tampa, FL)

Multiple redundant acoustic cues can contribute to the perception of a single class of speech sounds. This study investigates the effect of spectral degradation on the discriminability and perceptual saliency of acoustic cues for phonetic identification of word-final fricative voicing in "loss" versus "laws," and possible changes that occur when low-frequency acoustic cues are restored. Three acoustic cues that contribute to the word-final /s/-/z/ contrast (first formant frequency [F1] offset, vowel-consonant duration ratio, and consonant voicing duration) were systematically varied in modified natural speech syllables. The first experiment measured listeners' ability to discriminate differences among stimuli within a single cue dimension. The second experiment examined the extent to which listeners make use of a given cue to label a syllable as "loss" versus "laws" when multiple cues are available. Normal-hearing listeners were presented with stimuli that were either unprocessed, processed with 8-channel noise-band vocoder to approximate CI spectral degradation, or low-pass (LP) filtered to simulate low-frequency residual hearing. They were tested in four listening conditions: unprocessed, vocoder alone, LP alone, and vocoder + LP where low-frequency fine-structure cues could enhance F1 perception and voicing cues. The impact of listening condition on discriminability and weighting of different acoustic cues will be discussed.

2aSC18. Defining spectral and temporal resolutions of information-bearing acoustic changes for understanding noise-vocoded sentences. Christian Stilp (Dept. of Psychol. and Brain Sci., Univ. of Louisville, 308 Life Sci. Bldg., Louisville, KY 40292, christian.stilp@louisville.edu) and Matthew Goupell (Dept. of Hearing and Speech Sci., Univ. of Maryland, College Park, MD)

Information-bearing acoustic changes (IBACs) in the speech signal are highly important for speech perception. This has been demonstrated for full-spectrum sentences using cochlea-scaled entropy (CSE; Stilp and Kluender, 2010 *PNAS*) and noise-vocoded sentences using an adapted metric (CSE_{CI}; Stilp *et al.*, 2013 *JASA*). While IBACs appear fundamental to speech perception most broadly, Stilp *et al.* tested a single set of vocoder parameters, obscuring the breadth and depth of perceptual reliance upon these acoustic changes. Here, TIMIT sentences were noise-vocoded with variable spectral resolution (4-24 spectral channels spanning 300–5000 Hz), variable temporal resolution (4–64 Hz amplitude envelope cutoff frequency), or combinations therein. High-CSE_{CI} or low-CSE_{CI} sentence intervals were replaced with speech-shaped noise. As spectral resolution decreased, IBACs became more important for sentence understanding, especially at 6–12 channels. At high spectral resolutions, performance nearly overcame replacement of low-CSE_{CI} intervals but not high-CSE_{CI} intervals. Importance of IBACs at different temporal resolutions was largely driven by overall intelligibility, suggesting CSE_{CI} has sufficient temporal resolution at low modulation frequencies. Data exploring spectral-temporal tradeoffs will also be presented. Peak-picking strategies in CIs select and stimulate channels according to their amplitudes, but results suggest additional perceptual benefit may be offered by encoding IBACs as well.

2aSC19. Sentence intelligibility during segmental interruption and masking by speech-modulated noise. Daniel Fogerty (Dept. of Commun. Sci. and Disord., Univ. of South Carolina, 1621 Greene St., Columbia, SC 29208, fogerty@sc.edu), Jayne B. Ahlstrom (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC), William J. Bologna (Dept. of Hearing and Speech Sci., Univ. of Maryland, Charleston, South Carolina), and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Amplitude modulation from a competing talker can interfere with a listener's ability to process informative speech cues. The current study investigated how noise modulated by the wideband temporal envelope of a single competing talker impacts the contribution of consonants and vowels to the intelligibility of the target sentence. High variability speech materials were used, including talkers from different dialect regions. Young normal-hearing, older normal-hearing, and older hearing-impaired listeners completed speech recognition tests. All listeners received spectrally shaped speech matched to their individual audiometric thresholds to ensure sufficient audibility. Preliminary results demonstrated similar performance among the listener groups. When the modulated masker was interrupted, performance in consonant and vowel conditions was similar. However, sentence intelligibility in continuous single-talker-modulated noise was higher when vowels in the target sentence were preserved, as compared to consonants. Thus, masker continuity may facilitate source segregation for vowels more than for consonants. A second experiment that varied masker modulation rate for younger adults with normal hearing showed that modulation rate had more effect on processing cues conveyed by consonants than by vowels. Implications for hearing-impaired and older adults will be discussed. [Work supported by grants from NIH/NIDCD and ASHA.]

2aSC20. Acoustic properties of multi-talker babble. Noah H. Silbert (Commun. Sci. & Disord., Univ. of Cincinnati, 3202 Eden Ave., 344 French East Bldg., Cincinnati, OH 45267, noah.silbert@uc.edu), Kenneth de Jong, Kirsten Regier, Aaron Albin (Linguist, Indiana Univ., Bloomington, IN), and Yen-Chen Hao (Modern Foreign Lang. and Literatures, Univ. of Tennessee, Knoxville, TN)

Multi-talker babble can function as an excellent masker for speech stimuli in perception experiments. It has a higher degree of ecological validity than other maskers (e.g., white noise, speech-shaped noise), as it is a type of noise that many listeners encounter on a regular basis in everyday life. In addition, maskers constructed from speech have, by definition, acoustic properties similar to that of the signal. While multi-talker babble is used extensively in speech perception research, relatively little work has been done on the fine-grained acoustic properties of multi-talker babble. We present analyses of a number of acoustic properties of multi-talker babble generated by randomly combining phonetically balanced utterances (e.g., amplitude modulation depth, amplitude modulation frequencies, spectral properties, and spectro-temporal variability). In order to gain a fuller understanding of the nature of multi-talker babble, we analyze how the acoustic properties of babble vary as a function of the number (2–20), gender, and native language (English vs. Mandarin) of the speakers constituting the babble components. Future extensions of this work will (a) focus on how these acoustic variables affect speech perception, and (b) provide the foundation for a web-based system for generating customized samples of multi-talker babble noise for speech perception researchers.

2aSC21. The relationship between fluency, intelligibility, and acceptability of non-native spoken English. Mengxi Lin (Linguist, Purdue Univ., West Lafayette, IN 47907, lin211@purdue.edu) and Alexander L. Francis (Speech, Lang. & Hearing Sci., Purdue Univ., West Lafayette, IN)

Non-native accented speech is typically less intelligible and less fluent than native speech, but it is unclear how these factors interact to influence perceived speech quality. To investigate this question, the speech of 20 non-native speakers of English varying in proficiency and native language was evaluated. Subjective measures of speech quality (listening effort, acceptability, and intelligibility) were compared to objective measures of word recognition by native listeners, and to acoustic measures of fluency and of segmental and suprasegmental properties related to intelligibility. Results showed that subjective quality measures were highly related to one another and to word recognition and were most strongly predicted by measures of fluency. Segmental and suprasegmental measures did not predict word recognition or subjective speech quality. There was also an interaction between the effects of proficiency and speaker's native language on word recognition, but this did not extend to subjective measures. Finally, listeners who first heard high-proficiency speakers gave overall lower subjective quality ratings but there was no interaction between proficiency and presentation order. Multivariate analyses suggest that factors related to speaking rate, including pause duration, have the greatest effect on measures of acceptability, intelligibility, and listening effort. [Work supported by Purdue Linguistics and Purdue Research Foundation.]

2aSC22. Speech intelligibility can improve rapidly during exposure to a novel acoustic environment. Sofie Aspeslagh (MRC/SCO Inst. of Hearing Res. – Scottish Section, Glasgow Royal Infirmary, 16 Alexandra Parade, Glasgow G31 2ER, United Kingdom, sofie@ihr.gla.ac.uk), D. Fraser Clark (Dept. of Computing, Univ. of the West of Scotland, Paisley, United Kingdom), Michael A. Akeroyd, and W. Owen Brimijoin (MRC/SCO Inst. of Hearing Res. – Scottish Section, Glasgow, United Kingdom)

In natural listening environments, the background noises and the acoustic spaces they occupy can vary greatly, both in their characteristics and in their impact on speech intelligibility. It has been suggested that listeners build up a statistical representation of ongoing noises; however, listeners can move around a room or move from one room to another, constantly changing both the background noises and the reverberation around them. If listeners are to make use of such statistics to aid in speech intelligibility, they must engage in a constant adaptation. We asked normal and hearing-impaired listeners to identify in real-time a stream of random target words in a contiguous series of novel acoustic environments lasting 13 s each, thus measuring the extent and time-course of changes in speech intelligibility that listeners experience in a new environment. The results from our task demonstrate that most listeners experience a rapid increase in speech intelligibility over several seconds of exposure to certain acoustic environments, but not to others. This suggests that there are classes or types of noises and reverberations that can be adapted to over a short time course and others that cannot. [Work supported by the MRC and the CSO.]

2aSC23. Sizing down the competition: Speaking style and word recognition. Kristin Van Engen (Psych., Washington Univ., 176 Landa St. #514, New Braunfels, Texas 78130, kj.vanengen@gmail.com)

Identifying words with many phonological neighbors is more challenging for older adults than for younger adults. This difference has been attributed to reductions in inhibitory control associated with aging, which impair older adults' ability to cope with competition from similar-sounding words (Sommers and Danielson, 1999). Many difficulties in speech identification can be alleviated, however, when speech is produced clearly (i.e., the style adopted naturally by speakers when they perceive that their interlocutors are having difficulty understanding them). The current study investigates whether the acoustic-phonetic enhancements of clear speech can also reduce the inhibitory challenge of word recognition. If so, it is predicted that listeners will receive a greater benefit from clear speech when identifying lexically difficult words (i.e., words with many neighbors) vs. lexically easy words (i.e., words with fewer neighbors). Younger and older adults performed a word-recognition task in noise. Results to date show that the clear speech benefit is indeed greater for lexically difficult words than for lexically easy words for both groups of listeners. This pattern of results suggests that clear speech reduces the inhibitory demands associated with word recognition by increasing the perceptual difference between phonological neighbors.

2aSC24. Subcritical width rectangular bands of vocoded speech reveal the nature of envelope processing. James A. Bashford, Richard M. Warren, and Peter W. Lenz (Psych., Univ. of Wisconsin-Milwaukee, PO Box 413, Milwaukee, WI 53201, bashford@uwm.edu)

Bandwidth requirements for temporal envelope processing were examined using sentences that were reduced to arrays of sixteen rectangular bands (4800 dB/octave rolloff) having center frequencies ranging from 250

Hz to 8000 Hz, spaced at 1/3-octave intervals, and having subcritical bandwidths ranging from 4% to 0.5%. Envelopes extracted from the speech bands, without smoothing, were used to modulate white noise or tones centered at the speech band center frequencies. When the bandwidth of the modulated signals matched that of the speech, tone-vocoded arrays were more intelligible than noise-vocoded arrays and less intelligible than parent speech-band arrays. However, doubling the bandwidths of tone-vocoded arrays, which passed the upper and lower modulation sidebands fully, increased intelligibility to that of the parent speech array. Moreover, when the widths of modulated noise-band arrays were expanded to either an ERBn or 1/3-octave, their intelligibility equaled that of the speech for parent bandwidths of 4% and 2% and greatly exceeded speech array intelligibilities for parent bandwidths of 1% (58% vs. 25%) and 0.5% (28% vs. 3%). These and other findings indicate that optimal temporal envelope processing of speech requires that envelope cues stimulate a majority of fibers comprising critical bands. [Work supported by NIH.]

2aSC25. What explains perceptual weighting strategies of children with CIs: Auditory sensitivity or language experience? Susan Nittrouer, Amanda Caldwell-Tarr, and Joanna H. Lowenstein (Ohio State Univ., 915 Olentangy River Rd., Ste. 4000, Columbus, OH 43212, nittrouer.1@osu.edu)

Cochlear implants (CIs) have tremendously improved speech perception for deaf children, but problems remain. To examine why, this study compared weighting strategies of children with CIs and children with normal hearing (NH), and asked if these strategies are explained solely by the degraded spectral representations they receive through their implants, or if diminished opportunity to hear the ambient language accounts for some of the effect, as well. One hundred 8-year-olds (49 with NH and 51 with CIs) were tested on four measures: (1) labeling of a final-voicing contrast with one duration and one formant-transition cue; (2) labeling of a fricative-place contrast with one stable spectral and one formant-transition cue; (3) duration discrimination; and (4) glide discrimination. Children with NH and CIs weighted the duration cue similarly, suggesting children with CIs have sufficient experience to acquire language-appropriate strategies when cues are salient. Differences in weighting of spectral cues (both stable and time-varying) were found, but were not entirely explained by auditory sensitivity. The conclusion was that more salient cues facilitate stronger weighting, but individuals differ in how salient cues need to be to capture perceptual attention. Stimulus familiarity (speech or non-speech) affects how reliably children attend to acoustic cues, as well.

2aSC26. A phonetic basis for the sonority of [X]. Sarah Bakst and Jonah Katz (Linguist, Univ. of California Berkeley, 1915 Bonita Ave., Studio A, Berkeley, CA 94704, bakst@berkeley.edu)

Although after voiceless stops French rhotics are realized as voiceless uvular fricatives [X], they pattern phonologically as high-sonority liquids and are the only obstruent allowed between a consonant and a vowel. This poses a problem for the sonority hierarchy. This experiment tests whether [X], which has apparent approximant-like formant structure (Yeou and Maeda, 1995) patterns like the fricative [f] or the approximant [l] in its ability to convey information in formant transitions from a preceding consonant. In an AX burst detection task, native English speakers heard syllables of the form ClV, CXV, and CfV (spoken by a native French speaker) with and without a burst. Pilot data (n = 8; 410 trials each) suggests participants are more likely to respond "same" for [X] and [l] trials than for [f] trials ($p < 0.001$), suggesting that [X] carries more redundant information from a preceding consonant than [f] does and thus behaves more like approximants than other fricatives do. This suggests that the sonority of [X] is grounded in acoustics and perception, rendering an abstract account of sonority unnecessary. This result predicts that other fricatives with long front cavities should also be able to function as high-sonority segments.

2aSC27. A listener-based account for dispersion effects in sound change. Thomas Denby (Linguist, Northwestern Univ., 1525 W Estes Ave., Apt. B2, Chicago, IL 60626, tdenby@u.northwestern.edu), Grant McGuire, and Jaye Padgett (Linguist, Univ. of California-Santa Cruz, Santa Cruz, CA)

Phonetic dispersion has been proposed as the driving force behind a number of closely related sound-change phenomena. Listener-based accounts of dispersion (Labov, 1994, 587; Wedel, 2006; Denby, 2013) posit that phonetically unambiguous productions influence future productions of the listener more than ambiguous productions. The mechanism that drives this is a filter by which ambiguous productions are not stored, and thus do not update the phonemic categories of the listener. In turn, they are not reflected in that listener's future productions. In a new experiment, subjects heard words in noise and were asked to identify them by responding using a keyboard, following Goldinger (1996). Stimuli were from monosyllabic stop-initial minimal pairs differing in initial voicing, e.g., pat/bat. Half of these pairs were unambiguous productions, while the stop-initial VOT of the other half were manipulated to be somewhat ambiguous. If subjects store ambiguous words normally, their accuracy should improve with every exposure. If however, they do not store ambiguous productions, their accuracy should improve less than it does for unambiguous productions. Using d' scores, a repeated-measures ANOVA confirmed differences in improvement for unambiguous and ambiguous conditions were significant. A follow-up study is being implemented to test and expand these results.

2aSC28. Evaluation of web speech and lab speech for automatic classification of prosody. Jonathan Howell (Montclair State Univ., 1 Normal Ave., Montclair, NJ 07043, howellj@montclair.edu)

This study evaluates the performance of two very different sources of speech data in prosodic classification tasks: naturally occurring speech collected from the web (e.g., podcasts) and speech elicited in a laboratory. Speech from the web contains useful variability from which to generalize, e.g., variations in speaker and context; however, this variation also includes much statistical "noise," which may obscure the correct generalizations, e.g., different dialects or overlapping speech and other acoustic artifacts. We trained machine learning algorithms (support vector machines and linear discriminant analysis) to detect prosodic prominence in utterances of the comparative construction, e.g., "than I did." From 217 web-harvested utterances and 394 lab-elicited utterances (16 items, 27 subjects), we extracted more than 300 acoustic values, including measures of duration, F0, F1, F2, intensity, amplitude, voice quality, and spectral tilt. Both the web and lab training sets yielded similarly high-accuracy classifiers. The best performing algorithm achieved 87.6% accuracy ($p < 0.05$) when trained on web data and tested on lab data and 90.6% accuracy ($p < 0.05$) when trained on lab data and tested on web data. Significance values were calculated using a permutation-achieved empirical distribution.

2aSC29. Effects of perceptual anchors on nasality ratings in speech. Kristine E. Galek (Speech Pathol. and Audiol., Univ. of Nevada, Reno, P.O. Box 193, Carnelian Bay, CA 96140, kegh70@gmail.com) and Thomas Watterson (Speech Pathol. and Audiol., Univ. of Nevada, Reno, Reno, NV)

To study the effects of perceptual anchors on nasality ratings in speech. Speech samples were obtained from 95 hypernasal children and 5 normal controls. Samples were randomized and duplicated (6 sets of 100 samples). Six listening groups ($N = 129$) rated nasality on a seven-point scale ("1" normal nasality to "7" severe hypernasality). Anchors were located at different points along the continuum for each group. A single anchor located at scale value "4" or at scale values "3" and "5" educed an assimilation of ratings. Anchors placed at scale values "1" and "7" educed a contrast effect in that the distribution of ratings shifted away from the anchor sites. A single anchor placed at scale value "7" educed a systematic regression of the distribution of ratings away from the anchor site. Three anchors placed at scale values "1", "4" and "7" educed a more even distribution of ratings across the entire scale than the other five anchored conditions. Three of the five anchor groups consistently rated the first sample of the rating task a scale value "2" (based on median findings) even though the first sample was a normal control sample. The degree of nasality in speech is influenced by perceptual anchors.

2aSC30. Locus of phonological deficits in adults with dyslexia. Stephanie N. Del Tufo, Joslynn Noyes, Rebecca Sylvia, Sarah Montanaro, and Rachel M. Theodore (Univ. of Connecticut, 850 Bolton Rd., Unit 1085, Storrs, CT 06269, stephanie.del_tufo@uconn.edu)

Dyslexia is a developmental disorder that has traditionally been viewed as the consequence of impaired phonology. However, recent evidence suggests that the phonological deficits observed in dyslexia may reflect phonetic impairment earlier in the processing stream. The goal of the current work is to test this hypothesis. We use a phoneme-monitoring task to evaluate the relative use of two sources of information for making phonemic decisions (e.g., deciding if "dog" begins with /b/ or /d/). One source comes from a prelexical analysis of the speech signal (phonetic information) and the other source comes from a postlexical analysis (phonological information). Research has shown that healthy listeners flexibly shift between these sources based on experience with the acoustic signal. Specifically, phonemic decisions for native speech reflect phonetic information whereas phonemic decisions for foreign-accented speech reflect the use of phonological information. The interpretation is that phonetic processing is difficult for the foreign-accented speech, leading to a greater reliance on a postlexical analysis. If adults with dyslexia have impairment in phonetic processing, then we predict that phonemic decisions will reflect the use of phonological information for both types of speech, suggesting that the native speech signal is being processed as if it were accented.

2aSC31. Auditory and phonetic contributions to the neural mechanisms underlying vowel perception. Jeremy Burnison (Neurosci. Graduate Program, Univ. of Kansas, 1200 Sunnyside Ave., 4115 Haworth, Lawrence, KS, jburnison@ku.edu) and Jonathan Brumberg (Speech-Language-Hearing, Univ. of Kansas, Lawrence, KS)

One aspect of speech perception is our ability to identify the vowel sounds of spoken language. Identification accuracy of acoustic vowel targets has been shown to decrease as the first two formant frequencies of target sounds become more similar, suggesting that a vowel discretization exists in the F1-F2 auditory perceptual space. This is supported by prior work showing neural processes during vowel perception reflect recognition of discrete phonemes. In this study, we investigated the differential contributions of phonetic and acoustic factors on vowel perception using an event-related potential (ERP) protocol. Subjects first identified synthesized acoustic presentations of American English monopthong vowel sounds by selecting one of eight representative hVd words. Next, we presented synthesized sounds along the trajectory between two neighboring vowels in a mismatched negativity (MMN) oddball paradigm with one exemplar vowel as the standard stimulus. Preliminary results show statistically significant MMN amplitudes attenuate with standard-deviant vowel similarity, but include an additional MMN amplitude reduction at 50% vowel identification accuracy based on the behavioral responses. These results further clarify the neurological processing of vowel perception as a combination of auditory and phonetic factors in which acoustic differences elicit graded MMN responses that are augmented by shifts across phonetic boundaries.

2aSC32. Consonant confusability and its relation to phonological similarity. Sameer ud Dowla Khan (Linguist, Reed College, 3203 SE Woodstock Boulevard, Portland, OR 97202, sameeruddowlakhan@gmail.com)

Gradient similarity avoidance patterns in Bengali echo reduplication suggest that the most similar consonants to /t/ are, in order, /t, tʰ, d, t̪, s, t̪ʰ, k, .../. Converted to confusability, this ranking predicts that aspiration is most confusable, followed by voicing, minor place, continuancy, major place, and sonority. To confirm this, 24 native speakers identified syllables masked with babble, noise, or nothing ("clear"). Results indicate that confusability reflects similarity as predicted by avoidance patterns. In clear speech, most errors involved voicing and aspiration. Other errors reflected dialect-specific alternations. Noise introduced the percept of a loud burst: fricatives were often heard as affricates, non-alveolar coronals as alveolars, and non-coronals as coronals. Babble largely resembled noise. This pattern suggests that voicing is the most confusable feature, followed by aspiration, minor place and continuancy, major place, and lastly sonority. This ranking seems Bengali-specific, as studies of English find place to be significantly more confusable than manner and voicing. These results suggest that at least for Bengali, phonological alternations and perceptual confusability are argued to be better representations of how speakers judge similarity, rather than patterns in the lexicon or metrics such as shared natural classes metric of Frisch *et al*. (2004).

2aSC33. Information structure guides prominence perception. Jason Bishop (City Univ. of New York, 2800 Victory Blvd., Staten Island, NY 10314, jason.bishop@csi.cuny.edu)

The present study investigates the effect of information structure on the perception of prosodic prominence in English. In particular, we probed for top-down effects related to the size of the focus constituent (broad VP focus versus narrow object focus) in simple subject-verb-object sentences using a naïve prosody "transcription" task. In this task, listeners heard the same productions of a sentence, but in different information structural (i.e., question) contexts, and provided self-report decisions about the prominence of words using a Likert scale. Two primary questions were asked. First, does information structural interpretation produce expectation-based prominence perception? In this case, it was predicted that the presence of focus would induce perceived prominence independent of the signal (via expectations based on experience with production patterns). Second, does information structural interpretation modulate signal-based prominence perception? In this case, it was predicted that the presence of focus would enhance sensitivity to signal-based cues (via the modulation of attentional resources). Results are presented that show evidence for both types of effects, demonstrating a multifaceted influence of sentence-level semantic/pragmatic meaning on the perception of the signal.

Session 2aSP

Signal Processing in Acoustics: Session in Honor of William M. Carey I

James Lynch, Chair
Woods Hole Oceanogr., MS # 11, Bigelow 203, Woods Hole, MA 02543

Chair's Introduction—8:25

Invited Paper

8:30

2aSP1. A scientific crossroad: Carey's influence in affecting shallow water acoustics research after the cold war. Mohsen Badiey (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu) and David P. Knobles (Appl. Res. Labs., Univ. of Texas at Austin, Austin, TX)

Before the end of cold war most of the U.S. Navy's underwater acoustics research was focused on deep water. However, in the late 1980s and early 1990s, a shift toward shallow water environments was emphasized. With a body of knowledge from Ewing, Worzel, and Pekeris to Weston, Bill Carey took it upon himself to test a set of realistic goals in understanding complicated shallow water acoustics problems. His major field experiments in 1988–1993 produced results directing the community to address the issue of bottom attenuation and array coherence in a much more serious manner. In addition, due to his work, the community also considered the important problem of the effects of the water column, such as those associated with internal waves, on the acoustic propagation in these regions. This paper presents a summary of Carey's research in shallow water acoustics in light of the broader picture of the community's progress and direction in this field.

Contributed Papers

8:45

2aSP2. Laboratory measurements of compressional and shear wave speed and attenuation in muddy sediments. Megan S. Ballard, Kevin M. Lee, and Thomas G. Muir (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

As demonstrated by William M. Carey's field measurements in Dodge Pond, muddy sediments are characterized by a slow compressional wave speed and a low compressional wave attenuation [Carey and Pierce, POMA (5), 7001, 2009]. To gain insight into the measured data, a theoretical treatment of muddy sediments, named the Card House theory [Pierce and Carey, POMA (5), 7002, 2009], was developed. According the theory, isomorphous substitution causes each mud platelet to carry a net negative charge, and the resulting electrical interaction between platelets is responsible for creating a card-house structure. In this work, we examine laboratory measurements of compressional and shear wave properties in mud. Compared to the Dodge Pond measurements, which were affected by the size and distribution of gas bubbles present in the mud, the volume of gas in the laboratory samples was reduced by applying a vacuum. The estimated compressional wave speed is consistent with predicted values for a relatively gas-free mud. The estimated shear wave speed compares favorably with predicted values from the Card House theory. The electrochemical basis of the Card House model and its acoustical implications are also investigated. [Work supported by ARL:UT IR&D.]

9:00

2aSP3. Bill Carey and Bob Urick. David Bradley (Penn State Univ., PO Box 30, State College, PA 16870, dlb25@psu.edu)

Principles of Underwater Sound, written by Robert J. Urick, and last published in 1983, has been a staple on everyone's bookshelf. Bill Carey took on the job of updating that book, with the premise of not changing what was written, but simply updating each chapter with new information obtained over the succeeding three decades. One of the primary reasons the text was so popular is its ease in reading and grasping the import of the data and simplified ideas presented and their respective use(s) in the sonar equation. Work continues with Bill's firm imprint on the next edition, which is the focus of this discussion.

Invited Papers

9:15

2aSP4. Environmental factors in shallow water active sonar design. Peter Cable (135 Four Mile River Rd., Old Lyme, CT 06371, petercable@att.net)

In 1991, when William Carey was at DARPA, he initiated an effort under the Adverse Environments Program to both establish and define the limits of low frequency active sonar performance in shallow water and littoral regions, and to demonstrate a system concept that could achieve limiting performance. In support of that objective sea tests were conducted in the West Florida Shelf (Gulf of Mexico), Northeast U.S. Continental Shelf and Korea Strait (Area Characterization Tests I, II, and III), each in ~100m deep water with sand-

silt bottoms under downward refracting sound conditions. The design, conduct and subsequent analyses of those tests, including consideration of broadband sound transmission (100 Hz–1 kHz), signal dispersion and coherence, reverberation, bottom scattering strength, and clutter, will be traced with particular emphasis on Bill's leadership role in the endeavor. The key results of the program were sufficiently robust to guide the design of tactically significant shallow water active sonar, which they ultimately did.

9:30

2aSP5. Model-based underwater signal processing—The Carey factor. James V. Candy (Eng., Lawrence Livermore National Security, PO Box 808, L-151, Livermore, CA 94551, tsoftware@aol.com) and Edmund J. Sullivan (Prometheus, Inc., Newport, RI)

The sustained encouragement and belief in the model-based approach to underwater processing has always been one of the favorite topics of Bill Carey's conversation. Not only by his direct encouragement, but also his contributions in the form of well-executed, well-controlled and well-documented experiments in the Hudson Canyon area off of the New Jersey coast that has become known as the best and most complete sets of oceanic data available for signal processors to apply their latest algorithms. It has become affectionately known as the "canonical" oceanic signal generator and often stated by many signal processors that "if your algorithm is not capable of performing well on the Carey Hudson Canyon data, then it is not worthy of pursuing it further." His experimental work is a major contribution to the underwater processing area. In this paper, we briefly discuss the Hudson Canyon data set and show the performance of a model-based processor that was applied to localize a source using a 23-element hydrophone array in shallow water.

Contributed Papers

9:45

2aSP6. Bill Carey and sound attenuation in marine sediments. Ross Chapman (Univ. of Victoria, 3800 Finnerty Rd., Victoria, BC V8P5C2, Canada, chapman@uvic.ca)

Experiments carried out by Bill Carey in the Hudson Canyon off the New Jersey coast provided a great wealth of data for the study of low frequency sound propagation in shallow water. His analysis of the transmission loss data, reported in a series of papers dating from the mid 1990s, indicated a non linear frequency dependence of sound attenuation in the sediment material. This work provided a large base of experimental evidence for the non linear dispersion predicted by the Biot theory of sound propagation in porous media, and it stimulated new studies on sound propagation in marine sediments by many researchers, including Bill Carey himself. This paper reviews the results obtained by Carey from the Hudson Canyon experiments and places them in the context of new results from more recent experiments using different techniques and observables. Analysis of results from the new work indicates that Carey's observation of non linear frequency dependence in sediment material in the Hudson Canyon applies to attenuation of sound in different types of marine sediments.

10:00–10:15 Break

10:15

2aSP7. Spatial variation of seabed acoustic bulk properties. Nicholas P. Chotiros, Marcia J. Isakson, James N. Piper, and Andrew R. McNeese (Univ. of Texas at Austin, PO Box 8029, Austin, TX 78713-8029, chotiros@arlut.utexas.edu)

The seabed is modeled as a poro-elastic medium with a rough interface. The spatial variation of bulk properties, along with the interface roughness, are important contributors to the acoustic scattering strength of the seabed. Their effects are often indistinguishable. While roughness may be measured directly, the variability in the bulk properties is more difficult to obtain. In a recent experiment over a sandy seabed off Panama City, FL, known as the target and reverberation experiment of 2013 (TREX13), the seabed roughness and the normal acoustic reflection loss were simultaneously measured using a laser profiler and a short range acoustic sounder deployed aboard a remotely operated vehicle (ROV). Using the measured roughness statistics, the fluctuations in acoustic reflection loss due to roughness were estimated. Subtracting the roughness contribution from the total measured reflection fluctuations, the component due to bulk property changes was estimated, from which the fluctuation in the bulk properties may be inverted. [Work supported by the Office of Naval Research, Ocean Acoustics Program.]

Invited Paper

10:30

2aSP8. In memory of Bill Carey. Cathy Ann Clark (Sensors & Sonar Systems, NUWCDIVNPT, 1176 Howell St., B1320, R457, Newport, RI 02841, cathy.clark@navy.mil)

Bill Carey has been a strong and influential mentor to many young and not-so-young acousticians and engineers. He had a profound effect on my career, encouraging me to publish my work and to attend Acoustical Society Meetings. His work in bottom acoustics, noise directionality, and surface effects, in particular, provided significant input to my research. He was also instrumental in nominating me for Fellowship in the Acoustical Society. I am deeply grateful and appreciate having an opportunity to remember him in this session.

Contributed Papers

10:45

2aSP9. Measuring seismic waves using a towed underwater acoustic array. Jon M. Collis (Colorado School of Mines, 1500 Illinois St., Golden, CO 80401, jcollis@mines.edu) and Allan D. Pierce (Retired, East Sandwich, MA)

This paper discusses the possibility of detecting shear and interface (Scholte) wave effects in the ocean using a towed hydrophone array. The shear field will be evanescent in the water and so may only be detected near to the ocean bottom interface. A benefit of measuring the acoustic field with a towed array is that a Hankel transform can be used to construct the horizontal wavenumber spectrum. If a shear or interface wave is measured, then it will be visible in the horizontal spectrum. The possibility of detecting the shear field will be strongly dependent on the shear wave speed in the sediment and this will also affect the detection of the more difficult to detect Scholte wave, which travels at about 90 % the shear wave velocity. The Scholte wave has a circular polarization where the shear wave may be vertical, horizontal, or a combination of the two polarities and may not be detectable for all frequency and source depth configurations. [Work supported by ONR.]

11:00

2aSP10. Bill Carey and the development of our understanding of ocean acoustic coherence. John A. Colosi (Dept. of Oceanogr., Naval Postgrad. School, 833 Dyer Rd., Monterey, CA 93943, jacolosi@nps.edu)

One of the many topics for which Bill Carey had an enormous passion was coherence. Unlike many researchers who elect to stay in either the observational or theoretical sides, Bill jumped into both areas with zeal, uncovering important navy data and working with the theorists of the day. This talk will summarize some of Bill's most seminal work in the area of coherence, and the talk will demonstrate how Bill's ideas live on in recent developments on the subject.

11:15

2aSP11. A brief history of the modeling of sound propagation in bubbly liquids. Craig N. Dolder, Preston S. Wilson, and Mark F. Hamilton (Mech. Eng. Dept. and Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, dolder@utexas.edu)

William M. Carey is well known for his interest in sound propagation through bubbly liquids. He was also a champion of re-attributing the low frequency effective medium model widely known as Wood's law to its original author Arnulph Mallock, who published a paper titled "The Damping of Sound by Frothy Liquids" in 1910. In the same spirit, this presentation will discuss the evolution of theories involving sound propagation through bubbly liquids over time from Mallock to modern day. Since bubble pulsations can exhibit strong nonlinearity, the presentation will conclude by reintroducing another often-overlooked modeling advance, at least in the western literature, that of Zabolotskya and Soluyan [Sov. Phys. Acoust. **13**, 254–256 (1967)] describing the nonlinear propagation of sound in bubbly liquids. [Work supported by ONR.]

Invited Paper

11:30

2aSP12. From Wood to Carey to Mallock: A review of Bill Carey's work associated with the Mallock-Wood equation and the acoustics of bubbly liquids and gas-bearing sediments. Preston S. Wilson, Craig N. Dolder (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712, pswilson@mail.utexas.edu), Ronald A. Roy (Dept. of Eng. Sci., The Univ. of Oxford, Oxford, United Kingdom), and Allan D. Pierce (Dept. of Mech. Eng., Boston Univ., Boston, MA)

One of Bill Carey's many scientific interests throughout his career was the acoustics of bubbly liquids. Many underwater acousticians know of Carey's work associated with bubble clouds and more recently, gas-bearing sediments, but Bill got his start with the subject earlier in his career, studying flow in a boiling water reactor while employed at Argonne National Laboratory. Here, the acoustic velocity of bubbly liquid is of interest because of the possibility of supersonic flow, at comparatively low flow rates, in the high-void-fraction mixture within the reactor's piping system. In this talk, an overview of Bill's work with the acoustics of bubbly liquids will be presented, including scattering from bubble clouds, and sound propagation within bubbly liquid and gas-bearing sediments. Finally, Bill's campaign to rename a famous equation (Wood's Equation) in honor of its forgotten originator (Mallock) will be reviewed. [Work supported by ONR.]

Contributed Paper

11:45

2aSP13. A dipole source in a free surface and its representation as a near-surface monopole. Richard B. Evans (College of Eng., Boston Univ., 99F Hugo Rd., North Stonington, CT 06359, rbevans@99main.com)

A delta function impact on a free surface creates a dipole radiation pattern. This idealized source is of interest in the study of underwater ambient noise, in connection with noise caused by breaking waves, spray, and rain. The radiation pattern, due to the delta function impact, is derived and identified with the partial derivative of a fundamental monopole solution, or Green's function. Underwater acoustical models usually employ a monopole source. The representation of the surface dipole by a near-surface monopole is, therefore, a convenient approximation. The approximation of the dipole by a near surface monopole is an application of numerical differentiation. The reason for a distance of one quarter of a wavelength in the finite difference is described.

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

W.C. Foiles, 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
BP America, 501 Westlake Park Boulevard, Houston, TX 77079

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
National Institute of Standards and Technology (NIST), 100 Bureau Drive, Stop 8220, Gaithersburg, MD 20899

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

P.J. Battenberg, U.S. Technical Advisor for IEC/TC 29, Electroacoustics
3M Personal Safety Division, Detection Solutions, 1060 Corporate Center Drive, Oconomowoc WI 53066

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 S1, which will be held on Monday, 5 May 2014 from 5:15 p.m. - 6:30 p.m.

The Standards Committee Plenary Group meeting will precede the meetings of the Accredited Standards Committees S3, S3/SC 1, and S12, which are scheduled to take place in the following sequence:
Tuesday, 6 May 2014 11:00 a.m.-12:30 p.m. ASC S12, Noise
Tuesday, 6 May 2014 2:00 p.m. - 3:30 p.m. ASC S3, Bioacoustics
Tuesday, 6 May 2014 3:45 p.m. - 5:00 p.m. ASC S3/SC 1, Animal Bioacoustics

Accredited Standards Committee S2, Mechanical Vibration and Shock, is not scheduled to meet in Providence.

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
W.C. Foiles, Co-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		
P.J. Battenberg, U.S. TA	**IEC/TC 29** Electroacoustics	ASC S1 and ASC S3

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 - that meeting will be held in conjunction with the Standards Plenary meeting at 9:15 a.m. on Tuesday, 6 May 2014.

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 AFTERNOON, 6 MAY 2014 555 A/B, 1:05 P.M. TO 5:10 P.M.

Session 2pAAa

Architectural Acoustics: Uncertainty in Describing Room Acoustics Properties II

Lily M. Wang, Cochair
Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, PKI 101A, 1110 S. 67th St., Omaha, NE 68182-0816

Ingo B. Witew, Cochair
Inst. of Tech. Acoust., RWTH Aachen Univ., Neustrasse 50, Aachen 52066, Germany

Chair's Introduction—1:05

Invited Papers

1:10

2pAAa1. Measuring speech intelligibility using impulse responses: Impact of the decay range on the speech transmission index.
Constant Hak (Bldg. Phys. and Services, Eindhoven Univ. of Technol., Den Dolech 2, Eindhoven 5600 MB, Netherlands, c.c.j.m.hak@tue.nl) and Remy Wenmaekers (Level Acoust., Eindhoven, Netherlands)

IEC 60268-16 describes how to measure the Speech Intelligibility Index STI and its simplified derivatives such as STITEL and STIPA, using two measurement techniques. The first and oldest technique is based on a set of modulated noise signals used as a stimulus. The second method uses impulse responses obtained from maximum length sequence (MLS) or swept sine stimuli, deconvolution and Schroeder's Modulation Transfer Function (MTF). This technique is gaining more and more ground through advancing technology. The ability to modify the background noise component of a measured impulse response is one of the advantages of this technique, but requires a certain minimum impulse response quality. A measure of the quality of an impulse response is its decay range. The influence of the impulse response decay range on the calculated STI value is investigated. As in a previous study on the ISO 3382-1 parameters, this is done by using the Impulse response to Noise Ratio INR as an estimator for the decay range. The result is a proposal for the minimum required decay range to accurately measure the STI, based on the Just Noticeable Difference JND and the INR.

1:30

2pAAa2. Uncertainties in speech transmission index measurements. Peter Mapp (Peter Mapp Assoc., Copford, Colchester CO6 1LG, United Kingdom, petermapp@petermapp.com)

A detailed study has been carried out into the uncertainties associated with Speech Transmission Index (STI) measurements. The uncertainties and measurement errors are shown to be either systematic or random in nature. Systematic errors were found to include limitations of the technique itself as well as uncertainties related to measurement hardware and software implementations. Systematic errors were found to be caused by a range of issues including measurement microphone properties, test signal generation and replay errors, talker loudspeaker directivity and frequency response variations, and measurement system algorithms. Some forms of digital signal processing are also shown to affect the measured result and were found to be highly dependent upon the nature of the test signal and its processing. Measurement uncertainties due to random errors were found to include out of band, high sound pressure level, low frequency modulations or overloading of the microphone preamplifier stages, the pseudo random nature of the STI test signal itself, and the sensitivity of some test signals, such as maximal length sequences, to short term environmental and acoustic changes. Changes and fluctuations in the background noise level during measurements were also found to be a significant cause of error and uncertainly.

1:50

2pAAa3. Evaluation and improvement of a model to predict the measurement uncertainty due to the directivity of room acoustical sound sources. Ingo B. Witew, Mark Mueller-Giebeler, and Michael Vorlaender (Inst. of Tech. Acoust., RWTH Aachen Univ., Neustrasse 50, Aachen 52066, Germany, ingo.witew@akustik.rwth-aachen.de)

With the aim to reduce the necessary efforts to empirically determine the uncertainty in room acoustical measurements, in previous work, a model was developed that can predict the uncertainty a directivity of a sound source introduces to a measurement. As part of the validation extensive series of scale measurements have been conducted. In this contribution, the predicted uncertainty based on simulations and the empiric data is compared to each other. The results were used to improve the model. Concluding it will be discussed whether the model is suitable for a reasonable measurement uncertainty discussion.

2:10

2pAAa4. Measurement repeatability of late lateral energy level and lateral energy fraction. 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)

Late lateral energy level (GLL) and lateral energy fraction (LF) are two room acoustics measures that have been shown to correlate with certain aspects of spatial impression in concert halls. The purpose of this study was to investigate the repeatability of GLL and LF measurements. A custom microphone stand was built that can be adjusted in each spatial dimension separately to allow for accurate and precise microphone placement. Room impulse responses (IRs) were measured at six receiver locations in a 2500-seat auditorium using two different methods to obtain the lateral energy IR: the beamformed dipole response from a spherical microphone array, and a studio-grade figure-of-eight microphone. Three sets of IR measurements were taken at each receiver location. In between sets, the microphone stand was removed and the various adjustment points were randomly repositioned. The stand was then replaced in the same position to re-measure the IRs. The variability between the measurements at each receiver location was found to be relatively low for GLL (the standard deviation ranged between 0.22 and 0.73 JNDs for the 125–1000 Hz sum), and higher for LF (the standard deviation ranged between 0.49 and 2.80 JNDs for the average over the 125–1000 Hz octave bands). [Work supported by NSF Grant 1302741.]

2:30

2pAAa5. Using narrowly defined hypotheses to extract meaningful results from broad data sets. Scott D. Pfeiffer (Threshold Acoust. LLC, 53 West Jackson Blvd., Ste. 815, Chicago, IL 60604, spfeiffer@thresholdacoustics.com)

Data sets come in many forms, from modeled or calculated results, to *in-situ* measures, to bodies of work that provide collected sets of evidence. The collective data set can often be overwhelming, and can provide conflicting results, or present a level of uncertainty in analysis. Narrowly defining the questions to ask of the data based on the strengths and weaknesses of each tool seemingly enables discovery of the needle(s) in the haystack. Listening experiences in live settings provide the hypotheses that steer the analysis utilizing all of the tools at our disposal. Experience with targeted analysis of very specific phenomena are discussed from recent projects.

2:50

2pAAa6. Uncertainty in room acoustics: A consultant's perspective. Benjamin Markham and Jonah Sacks (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

Faced with time and budget constraints on most projects, acoustical consultants make a series of choices when asked to evaluate the acoustics of an existing space. Measure the impulse response of the room with sophisticated technology (MLS, sine-sweeps, etc.), or pop a balloon? Use a dodecahedral loudspeaker (or several), a more standard portable amplified loudspeaker, or the house sound system (or pop a balloon)? Use a single omni-directional microphone, a binaural dummy head, or a B-format soundfield microphone? How many source locations, and how many receiver locations? Whither measure? There is a growing body of literature that identifies uncertainty and inconsistency in reverberation time and other room acoustics metrics and the tools used to obtain them. The authors will add modestly to this evidence from consulting experience, and then focus on how some consultants in architectural acoustics choose what data to gather, how to gather it, and to what extent. Considerations of time, budget, and logistics factor in those choices, as do determinations regarding the essence of the client's needs and concerns, subjective listening to determine what measurements could be most useful, and our own expectations for how we might use the measurements we make, now and in the future.

3:10–3:25 Break

3:25

2pAAa7. Evaluation of acoustic diversity of religious buildings; case study from churches and mosques in Turkey. Filiz B. Kocyigit (Architecture, Atilim Univ., Kizilcasar Mah. Incek, Ankara 06560, Turkey, filizbk@gmail.com), Gamze Akbaş, Gokhan Ozal, Can Yerli (Fine Art Design and Architecture, Atilim Univ., Ankara, Turkey), Meral Gunduvan (Fine Art Design and Architecture, Atilim Univerity, Ankara, Turkey), and Feyyaz Demirer (English Education, Harran Univ., Urfa, Turkey)

In this study, by analyzing the reverberation time and noise insulation features of religious structures, effects of differences on the forms, materials, and functions on interior acoustics are discussed. For this purpose, calculations were made working on the projects of the selected sample mosques and churches built by different materials and their interior reverberation times were measured and compared. Two different types of areas were compared using RT60 reverberation time calculations and T30 and T20 frequency band analysis measurement systems. In the comparison, interior functions are evaluated. In this evaluation, it was observed that in mosques since there is excessive speaking, medium frequency resulting from people was more dense whereas in churches high frequency was also involved with rite taking place alongside the medium frequency resulting from talking. Therefore, interior frequency band width was observed to increase. Selected churches have cross and basilic type plan and mosques have square type plan. It is observed that selected traditional mosque roofs have circular forms and churches have sharp forms. Hereby variations of the acoustic features of the sites with different forms and functions used for similar purposes were compared.

3:40

2pAAa8. The development and analysis of a large variable acoustics space. Jay Bliefnick (Acoust., Columbia College, 5001 River Rd., Apt. 1S, Schiller Park, IL 60176, jay.bliefnick@loop.colum.edu), Andrew Hulva, and Dominique Cheenne (Acoust., Columbia College, Chicago, IL)

A new, large-scale variable acoustics space has recently been added to the Audio Arts & Acoustics department at Columbia College Chicago. Built within the current Motion Capture studio, this facility will provide students and faculty the ability to perform tests in an acoustically controlled environment, without the influence of small-room effects. The construction involved the creation of nearly 300 2 in. x 2 in. reversible boxes: one side diffusive and the other absorptive. These line three full walls of the space at a height of 10 in., totaling ~1200 ft^2 of acoustically adjustable surface area. This allows the room to convert from a very absorptive space, to one that is much more acoustically active. Multiple specular reflector panels are also available for the creation of "hot spots," allowing for even more diverse applications. This study focused on the construction and initial testing of this innovative new space. To analyze the effectiveness of the additions, frequency, time, and reverberation responses for the entire room were sampled in a variety of configurations: fully absorptive, fully diffusive, empty, etc. These objective metrics were then analyzed against perceptual data to determine the correlation between what could be measured and what could be heard.

3:55

2pAAa9. Evaluating methods of acoustic analysis in a small listening room. Jennifer Levins (SoundSense, LLC, 2669 E Thompson St., Philadelphia, PA 19125, jenlevins@gmail.com)

Although common, octave band analysis of decay times provides limited information on the behavior of sound in a room. In spaces for critical listening, additional analysis is required in the design phase. Even in existing spaces, measurements of reverberation time do not always reveal room anomalies such as echoes and standing waves. The acoustic experience of the space is not always indicated by reverberation time. In rooms for critical listening, additional methodologies such as modal analysis, ray tracing, and early reflection times are used to determine additional information about a room's acoustic characteristics. A case study will be presented to demonstrate the applicability of these methods in small listening rooms.

4:10

2pAAa10. Experimental verification of computer modeled loudspeaker sound level performance based on excitation signal. David S. Woolworth (Oxford Acoust., 356 CR 102, Oxford, MS 38655, dave@oxfordacoustics.com) and Vahid Naderyan (Phys., Univ. of MS, University, MS)

An investigation of a loudspeaker sound level performance in a simple geometry to verify the EASE sound level definition comparing pink noise and multi-tone signal utilizing field tests of loudspeakers.

4:25

2pAAa11. Efficient computational modeling of Platonic solid loudspeaker directivities. Jeshua H. Mortensen and Timothy W. Leishman (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, meako490@gmail.com)

Dodecahedron loudspeakers are commonly used in architectural acoustics measurements as quasi-omnidirectional sources of sound. Other Platonic solid loudspeakers may also be used for this purpose, but the geometrical properties that optimize their omnidirectional behaviors are not well understood. Because of the difficulties in constructing large numbers of loudspeakers and boundary element models for related investigations, a computational method has been developed for the MATLAB environment to rapidly predict radiated fields and observe general directivity trends as geometrical properties vary. The method is based on related spherical enclosure geometries. It enables one to easily assess the effects of altered driver diameters, positions, numbers, vibrational patterns, and enclosure volumes. This presentation discusses the tool and presents several computational findings. Its output is presented as animated frequency-dependent balloon plots and area-weighted spatial standard deviations. The results are found to agree well with similar predictions for actual Platonic solid geometries from the boundary element method and experimental measurements.

4:40

2pAAa12. A closer look at the Hopkins-Stryker equation. Timothy W. Leishman and Zachary R. Jensen (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., N247 ESC, Provo, UT 84602, twleishman@byu.edu)

The Hopkins-Stryker equation has long been used to represent sound fields in semi-reverberant rooms. However, its implementation could be improved if users were more familiar with its origins, assumptions, and potential applications. The directivity factor, distance from the acoustic center, room constant, and locally averaged energy density are all key elements of the equation that merit special attention. This presentation explores these quantities and their theoretical underpinnings. It also introduces the use of generalized energy density as a means of simplifying averaging requirements. Selected numerical examples serve as illustrations to clarify the concepts.

4:55

2pAAa13. A two-point method for direct measurement of the room constant. Zachary R. Jensen and Timothy W. Leishman (Acoust. Res. Group, Dept. of Phys. and Astronomy, Brigham Young Univ., ESC, Provo, UT 84604, zjens1@gmail.com)

The room constant is a key frequency-dependent value that is widely used to characterize reverberant fields. It is typically estimated from room boundary properties, viz., total surface area and average absorption coefficient. Unfortunately, these properties are often difficult to ascertain with sufficient accuracy. While reverberation times may be adequately measured using modern methods, the effective surface areas and volumes of many practical rooms are elusive. Furthermore, several formulations for the room constant exist without general agreement as to their best usage. This presentation introduces a two-point energy-based method that enables acousticians to feasibly measure the room constant without knowledge of the room volume, surface area, or average absorption coefficient. Numerical simulations and measurements illustrate the benefits of the approach. Resulting values are compared with those approximated using common formulations for validation and clarification.

Session 2pAAb

Architectural Acoustics and National Council of Acoustical Consultants: Student Design Competition (Poster Session)

Norman H. Philipp, Chair
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 2014 Student Design Competition that will be professionally judged at this meeting. The 2014 design competition involves the design of a fine arts building for a high school of moderate size primarily for a school's strong opera program. The submitted designs will be judged by a panel of professional architects and acoustical consultants. An award of USD$1,250 will be made to the submitter(s) of the design judged "first honors." Four awards of UDS$700 each will be made to the submitters of four entries judged "commendation."

TUESDAY AFTERNOON, 6 MAY 2014 554 A/B, 1:00 P.M. TO 5:00 P.M.

Session 2pAB

Animal bioacoustics: Acoustics as a Tool for Population Structure I

Kathleen Stafford, Cochair
Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105

Shannon Rankin, Cochair
Southwest Fisheries Science Ctr., 8901 La Jolla Shores Dr., La Jolla, CA 92037

Chair's Introduction—1:00

Invited Papers

1:05

2pAB1. Geographic variation in acoustic signals of freshwater fishes. Carol Johnston (Fisheries, Fish Biodiversity Lab., Auburn Univ., Swingle Hall, Auburn, AL 36830, johnsc5@auburn.edu), Catherine Phillips (US Fish and Wildlife Service, Panama City, FL), and Patty Noel (Fisheries, Auburn Univ., Auburn, AL)

Although well studied in other taxa, geographic variation in signal structure has been poorly studied in fishes. Signal divergence in marine fishes depends on larval form; pelagic larvae disperse more widely than demersal forms, limiting opportunities for isolation and subsequent divergence. Freshwater fishes, especially those restricted to headwater habitat, are isolated by drainage. There are many examples of species radiations in groups of North American freshwater fishes within drainage networks, including darters, minnows, and catfishes, some of which are restricted to single streams. Our data demonstrate divergence of acoustic signals among populations of stream and riverine fishes at multiple scales, and often in the absence of apparent morphological variation. Two model species, Longear Sunfish and Whitetail Shiner, differed in the temporal components of calls, while darter and sturgeon models showed variation in both temporal and spectral call components. In the case of the sturgeon, the populations were genetically distinct. Furthermore, data for Whitetail Shiner suggest that calls associated with courtship were strongly associated with geographic isolation, while divergence in those characteristics associated with aggression may be driven by genetic drift. We suggest that variation in acoustic signal structure may be common in freshwater fishes and discuss implications for mate choice.

1:25

2pAB2. Sources of acoustic variation in the advertisement vocalizations of Neotropical singing mice. Bret Pasch (Dept. of Integrative Biology, Univ. of Texas at Austin, Austin, TX 78712, bpasch@utexas.edu), Polly Campbell (Dept. of Zoology, Oklahoma State Univ., Stillwater, OK), Mustafa Z. Abbasi, Preston S. Wilson (Mech. Eng., Univ. of Texas at Austin, Austin, TX), Steven M. Phelps, and Michael J. Ryan (Dept. of Integrative Biology, Univ. of Texas at Austin, Austin, TX)

Patterns of variation in communication systems provide important insight into the processes that shape phenotypic evolution. Although studies in anurans, birds, and insects indicate that diverse selective and stochastic forces influence acoustic signals, factors that shape variation in mammalian vocalizations are poorly understood. Neotropical singing mice (*Scotinomys*) are diurnal, insectivorous rodents distributed throughout montane cloud forests of Middle America. Males commonly emit species-specific advertisement vocalizations that are used in mate attraction and male-male aggression. To explore factors contributing to vocal variation, we summarize data from a diversity of studies at disparate scales and levels of analysis. We highlight the importance of genetic drift in shaping population differentiation, the role of androgens in modulating the performance of physically challenging displays, the influence of social context in shaping posture and vocal parameters, and the impact of the ambient environment in affecting sound propagation. Neotropical singing mice are emerging as an important model that enables us to draw parallels to vocal communication systems in traditionally more tractable species.

1:45

2pAB3. Vocal diversity and taxonomy of white-cheeked crested gibbons. Julia Ruppell (Biology, Pacific Univ., 3920 SW Alice St., Portland, OR 97219, Ruppell@pacificu.edu)

Previous research suggests that gibbon song repertoire is genetically determined and song characteristics are useful for assessing systematic relationships. The taxonomy and distribution of crested gibbons (genus *Nomascus*) had not been studied previously. In addition, crested gibbons face several threats to extinction such as habitat loss, the pet trade, the domestic and international trade of wildlife, and unsustainable harvest of wildlife for subsistence consumption. Additionally, rice field expansion and poaching pose significant threats to gibbons and their habitats, especially because most populations are very small. I studied vocal diversity among different wild populations of *Nomascus* in Vietnam and Laos to assess their taxonomic relationships and to examine whether their vocal patterns correspond to forms previously described. Linear discriminant analysis, classification trees, and multidimensional scaling revealed distinct populations based on song characteristics and species (or subspecies) boundary locations were recognized. In addition, population sizes in different areas were estimated based on vocal analysis. The recognition of previously unknown diversity within *Nomascus* and ability to locate potential species boundaries aided in implementation and adaptation of gibbon conservation strategies and the development of gibbon management plans for protected areas. In addition, this project improved the monitoring of a poorly known and understudied ape, by working with provincial staff and local people.

2:05

2pAB4. Behavioral and phylogenetic differentiation in a potential cryptic species complex, the canyon treefrog. Katy Klymus, Carl Gerhardt, and Sarah Humfeld (Univ. of Missouri, 302 ABNR, Columbia, MO 65201, klymusk@missouri.edu)

Detection of genetic and behavioral diversity within morphologically similar species has led to the discovery of cryptic species complexes. We tested the hypothesis that the canyon treefrog (*Hyla arenicolor*) may consist of cryptic species by examining mate-attraction signals among highly divergent lineages defined by mitochondrial DNA (mtDNA). Unexpectedly, calls exhibited little variation among the three U.S. lineages despite large mtDNA sequence divergences. We re-analyzed intraspecific and interspecific phylogenetic relationships by sequencing both mitochondrial and nuclear genetic markers among populations and a closely related, but morphologically and behaviorally different species, the Arizona treefrog (*H. wrightorum*). Discordance between mitochondrial and nuclear datasets suggests multiple instances of introgression of *H. wrightorum's* mitochondrial genome into populations of *H. arenicolor*. Furthermore, intraspecific population structure based on nuclear markers shows better congruence with patterns of call variation than population structure based on the mitochondrial dataset. Although the U.S. lineages do not appear to represent cryptic species, Mexican lineages do show biologically relevant call differences as assessed through female preference tests. Our results suggest that call variation can indicate genetic structure of populations; however, a multilocus approach should be used in defining genetic structure, as using only mtDNA may lead to erroneous conclusions.

Contributed Papers

2:25

2pAB5. Population structure of humpback whales in the western and central South Pacific Ocean determined by vocal cultural exchange. Ellen C. Garland (National Marine Mammal Lab., AFSC/NOAA, AFSC/NOAA, 7600 Sand Point Way NE, Seattle, WA 98115, Ellen.Garland@noaa.gov), Michael J. Noad (Cetacean Ecology and Acoust. Lab., School of Veterinary Sci., Univ. of Queensland, Gatton, QLD, Australia), Anne W. Goldizen (School of Biological Sci., Univ. of Queensland, St. Lucia, QLD, Australia), Matthew S. Lilley (SecuritEase Int.., Petone, New Zealand), Melinda L. Rekdahl (Cetacean Ecology and Acoust. Lab., School of Veterinary Sci., Univ. of Queensland, Gatton, QLD, Australia), Claire Garrigue (Opûration Cûtacûs, Noumea, New Caledonia), Rochelle Constantine (School of Biological Sci., The Univ. of Auckland, Auckland, New Zealand), Nan Daeschler Hauser (Cook Islands Whale Res., Avarua, Rarotonga, Cook Islands), M. Michael Poole (Marine Mammal Res. Program, Maharepa, Moorea, French Polynesia), and Jooke Robbins (Provincetown Ctr. for Coastal Studies, Provincetown, MA)

Male humpback whales (*Megaptera novaeangliae*) produce a continually evolving vocal sexual display, or "song," which is shared by all males within a population. The rapid cultural transmission of this display between distinct but interconnected populations within the western and central South Pacific region presents a unique opportunity to investigate population connectivity based on a vocal display. Quantitative analyses were conducted on eleven years of data to investigate vocal groupings based on song types shared between populations, to produce an acoustically derived population structure for the region. Four distinct vocal groupings resulted; the western group contained a single population, off eastern Australia, the central group was comprised of whales around New Caledonia, Tonga and American Samoa, and finally the whales of the eastern region were split into two groups, one around the Cook Islands and the other in the waters of French Polynesia. These groupings broadly agree with results obtained using genetic and photo-identification methods, and confirm that humpback whales are likely to form separate breeding populations rather than panmictic subpopulations. This study demonstrates the utility of using culturally transmitted vocal patterns as a way of defining populations, at least in this species.

2:40

2pAB6. Using passive acoustics to investigate seasonal and diel trends in acoustic behavior of North Atlantic right whales (*Eubalaena glacialis*). Leanna P. Matthews, Jessica A. McCordic, and Susan E. Parks (Biology, Syracuse Univ., 227 Life Sci., 107 College Pl., Syracuse, NY 13244, lematthe@syr.edu)

The North Atlantic right whale (*Eubalaena glacialis*), an endangered baleen whale species, produces a variety of stereotyped acoustic signals. One signal, the "gunshot" sound, has only been recorded from adult males and is thought to function for reproduction, either as advertisement for females or an agonistic signal toward other males. This study uses remote acoustic monitoring to analyze the presence of gunshots over a two-year period at two sites on the Scotian Shelf to determine if there is evidence that right whales use these locations for breeding activities. Seasonal analyses at both locations indicate that gunshot production is highly seasonal, with an increase in autumn. One site had significantly more gunshot sounds and exhibited a clear diel trend in signal production. The other site also showed a seasonal increase during autumn, but did not show any significant diel trends. This site difference indicates variation either in the number or the behavior of whales at each location. The timing of the observed seasonality in gunshot production is consistent with the current understanding of right whale breeding season, and these results demonstrate that detection of gunshots with remote acoustic monitoring can be a reliable way to track seasonal mating activities.

2:55–3:15 Break

3:15

2pAB7. Variation in the acoustic behavior of right whale mother-calf pairs. Susan Parks (Dept. of Biology, Syracuse Univ., 107 College Pl., Syracuse, NY 13244, sparks@syr.edu), Lisa Conger (NOAA Fisheries, Northeast Fisheries Sci. Ctr., Woods Hole, MA), Dana Cusano (Dept. of Biology, Syracuse Univ., Syracuse, NY), and Sofie Van Parijs (NOAA Fisheries, Northeast Fisheries Sci. Ctr., Woods Hole, MA)

North Atlantic right whale mother-calf pairs are a critical segment of the population for recovery of this endangered species and therefore their protection is paramount. Passive acoustics, including the development of real-time buoys, has increasingly played a role in the detection of right whale presence in high vessel traffic areas. The vocal behavior of North Atlantic right whale mothers and their young calves has not been well described and may not be well represented by calls produced by other individuals in the population that are commonly utilized for passive acoustic monitoring. Therefore, it is critical to determine the call types and rates of sound production by mother-calf pairs to assess the efficacy of passive acoustic monitoring for their detection. We conducted behavioral focal follows coupled with acoustic recording of right whale mother calf pairs off the coast of Florida and Georgia in January–March, Cape Cod Bay in April, and the Bay of Fundy in August–September from 2011 to 2014. Results show modifications in both call structure and call rate with increasing calf maturity and independence. These data are necessary to better utilize passive acoustic monitoring for management purposes in this species.

3:30

2pAB8. Western and central North Atlantic fin whale (*Balaenoptera physalus*) stock structure assessed using geographic song variations. Julien Delarue (JASCO Appl. Sci., 202 - 32 Troop Ave., Dartmouth, NS B3B 1Z1, Canada, julien.delarue@jasco.com), Robert Dziak, David Mellinger (PMEL/NOAA, Newport, OR), Jack Lawson (Fisheries and Oceans Canada, St. John, NF, Canada), Hilary Moors-Murphy (Fisheries and Oceans Canada, Dartmouth, NS, Canada), Yvan Simard (Fisheries and Oceans Canada, Mont-Joli, QC, Canada), and Kathleen Stafford (Appl. Phys. Lab, Univ. of Washington, Seattle, WA)

Variations in calls or songs between areas are increasingly acknowledged as a way to assess stock structure. We present the results of an analysis of 163 fin whale (FW) songs recorded in seven areas of the North Atlantic (NA): Irminger Sea, Davis Strait, Grand Banks of Newfoundland, Southern Newfoundland, Gulf of St. Lawrence, eastern Scotian Shelf, and the waters off Delaware Bay. Song measurements included inter-note intervals (INI), notes' peak frequency and bandwidth and note type (classic, backbeat, and high-frequency) proportion. Seasonal patterns of classic-classic INI provided the highest level of differentiation between areas and revealed the existence of six acoustic stocks. Classification trees revealed that other parameters distinguished between regions over larger spatial scales, grouping some of the recording areas together. These results suggest that (1) there are four distinct stocks in the western NA; (2) the range of a presumed central NA stock includes southwestern Iceland, both sides of Greenland and appears to extend south along the Mid-Atlantic Ridge, at least in recent years; (3) two stocks are present off West Greenland. These results bring new information on potential FW stock delineations in the NA. The latter will be compared to those derived using other stock assessment metrics.

3:45

2pAB9. Do spectral features of Risso's dolphin echolocation clicks vary geographically? Melissa Soldevilla, Lance Garrison (NOAA Southeast Fisheries Sci. Ctr., 75 Virginia Beach Dr., Miami, FL 33149, melissa.soldevilla@noaa.gov), Simone Baumann-Pickering (Scripps Inst. of Oceanogr., UCSD, La Jolla, CA), Danielle Cholewiak, Sofie Van Parijs (NOAA Northeast Fisheries Sci. Ctr., Woods Hole, MA), Lynne Hodge, Andrew Read (Duke Univ. Marine Lab., Beaufort, NC), Erin Oleson (NOAA Pacific Islands Fisheries Sci. Ctr., Honolulu, HI), and Shannon Rankin (NOAA Southwest Fisheries Sci. Ctr., La Jolla, CA)

The ability to classify odontocetes to species and population from acoustic recordings leads to improvements in stock identification, abundance and density estimation, and habitat-based density modeling, which are crucial

for conservation and management. Risso's dolphins off Southern California have distinctive peaks and valleys in their echolocation clicks, which allow researchers to easily distinguish them from other species in passive acoustic recordings. However, Risso's dolphin echolocation clicks from other geographic areas have not been described and it remains unknown whether they have similarly distinctive click spectra and whether stocks are acoustically distinct. We investigate the potential for using acoustics to identify populations by quantifying the acoustic structure of Risso's dolphin echolocation clicks recorded over wide-ranging geographic regions including the U.S. waters of the North Atlantic Ocean (north and south of Cape Hatteras), Gulf of Mexico, and North Pacific Ocean (Eastern Tropical Pacific and Southern California). Several distinctive peak and valley patterns are found and we evaluate these in light of variability within individuals, groups, and regions as these acoustic differences may indicate differences in age, sex, or stock composition, foraging behavior, or ambient noise environment of the dolphin schools.

4:00

2pAB10. Counting odontocetes from click train detections using multiple independent autonomous acoustic sensors. James A. Theriault (Ocean and Ecosystem Sci., Bedford Inst. of Oceanogr. , Dartmouth, NS, Canada), Craig Sheppard, Joey Hood (Akoostix, Inc., 10 Akerley Blvd., #12, Dartmouth, NS B3B 1J4, Canada, jhood@akoostix.com), Hilary B. Moors-Murphy (Ocean and Ecosystem Sci., Bedford Inst. of Oceanogr. , Dartmouth, NS, Canada), and Matthew Coffin (Akoostix, Inc., Dartmouuth, NS, Canada)

Passive acoustic monitoring (PAM) is often suggested as an effective technology to mitigate impacts from anthropogenic activities; however, the ability to reliably and efficiently detect, locate, and count cetaceans using PAM is still in development. One particularly useful application of PAM is species density estimation, which requires an estimate of the number of individuals involved in a detection event. Efforts have been undertaken to develop methods to directly count the number of vocalizing animals during acoustic detection events. For odontocetes, discrete clicks are almost indistinguishable between individuals, making it more difficult to determine the number of vocalizing animals as the number increases. Using recordings from multiple closely spaced ($\approx$200 m) GuardBuoy sensors deployed on the Canadian Scotian Shelf, cross-sensor correlograms were produced to estimate the number of individual sperm whales, and, as a more challenging case, the number of vocalizing delphinids. Using feature-based multipath reflection discrimination, the raw time series were reduced to a synthetic time series of binary click detections with the multipath arrivals removed. The synthesized click-detection time series were used for the cross-sensor correlograms to generate improved estimates of the number of vocalizing animals as compared with using the raw time series.

4:15

2pAB11. The acoustic structure of whistles as a tool for identifying evolutionary units in dolphins. Elena Papale, Marta Azzolin (Dept. of Life Sci. and Systems Biology, Univ. of Torino, via accademia albertina 13, Torino 10123, Italy, elena.papale@unito.it), Irma Cascao (Centro do Instituto do Mar (IMAR) da Universidade dos Açores, Departamento de Oceanografia e Pescas, Universidade dos Açores, Horta, Portugal), Alexandre Gannier (Groupe de Recherche sur les Cétacés, Groupe de Recherche sur les Cétacés, Antibes, France), Marc O. Lammers (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Kanehone, HI), Julie N. Oswald (Bio-Waves, Bio-Waves, Encinitas, CA), Monica Perez-Gil (Society for the Study of Cetaceans in the Canary Archipelago, Puerto Calero, Spain), Monica Silva (Centro do Instituto do Mar (IMAR) da Universidade dos Açores, Departamento de Oceanografia e Pescas, Universidade dos Açores, Horta, Portugal), and Cristina Giacoma (Dept. of LIfe Sci. and Systems Biology, Univ. of Torino, Torino, Italy)

Acoustic signals are expressions of phenotypic diversity and their variation could provide important information on differentiation patterns within species. Due to a number of selective pressures acting on signals, the contribution of genetic drift is often complex to outline. This study aims at evaluating if an examination of the acoustic structure of communication signals can allow the identification of evolutionary units in species capable of vocal learning. We quantified and compared parameters of whistles emitted by three dolphin species (*Stenella coeruleoalba*, *Delphinus delphis*, and *Tursiops truncatus*) to examine the hypothesis that acoustic signals permit the recognition of differentiation between populations from the Atlantic Ocean and the Mediterranean Sea. In the three species, recordings were correctly assigned to their basin of origin with a percentage higher than 82% by DFA. Frequency parameters were the most stable within each species. Where gene flow has been shown, i.e., within Atlantic Ocean, significant differences were found principally in modulation parameters. We hypothesize that these parameters are influenced by social and behavioral factors and that similar ecological conditions led to convergent acoustic features. Results of this study suggest that is it possible to recognize evolutionary units based on acoustic data.

4:30

2pAB12. Long-term acoustic surveying of bats. Annemarie Surlykke, Tûrur Andreassen (Biology, Univ. of Southern Denmark, Campusvej 55, Odense DK-5230, Denmark, ams@biology.sdu.dk), and John Hallam (Maersk-McKinney Moller Inst., Univ. of Southern Denmark, Odense, Denmark)

Increasing concern about decline in biodiversity has created a demand for population surveys. Long-term unmanned automatic monitoring may provide unique unbiased data from a whole season, but the large amount of data presents serious challenges for automatic processing. A two-month study of echolocating bats at 500 kHz sampling rate provided 236 GiB of data at full bandwidth. We used a Support Vector Machine (SVM) classier based on a combination of temporal and spectral analyses to classify events into bat calls and non-bat events. Duration, energy, bandwidth, and entropy were used to identify bat calls and reject short noise pulses, e.g., from rain. The SVM classifier reduced our dataset to 162 MiB of candidate bat calls with an estimated accuracy of 96% for dry nights and 70% when it was raining. The automatic survey revealed correlation between bat activity and rain, temperature, and sunset/sunrise. There were calls from two species new to the area, as well as an unexpected abundance of social calls. Future applications aim at higher accuracy in classifying bat calls and using trajectory-tracking to determine flight paths to correct for the bias toward loud bats inherent in acoustic surveying.

4:45

2pAB13. Small-scale soundscapes over coral reef ecosystems: Latitudinal variation in the Northwestern Hawaiian Islands. Simon E. Freeman (Marine Physical Lab., Scripps Inst. of Oceanogr., 3725 Miramar St. Apt. C, La Jolla, CA 92037, sfreeman@ucsd.edu), Lauren A. Freeman (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA), Marc O. Lammers (Oceanwide Sci. Inst., Honolulu, HI), and Michael J. Buckingham (Marine Physical Lab., Scripps Inst. of Oceanogr., San Diego, CA)

Two-dimensional seafloor "maps" of near-field ambient sound produced by biological sources in a coral reef environment were obtained from four spur-and-groove shallow water reef environments along a latitudinal gradient in the Papahanaumokuakea Marine National Monument, Northwestern Hawaiian Islands. Acoustic data were collected in conjunction with SCUBA based ecological surveys in an effort to establish correlation between components of the acoustic field and the ecological state of each field site. Simultaneous acoustic measurements and remote underwater photographs taken during the day and at night allowed for comparisons of biological activity with recordings over time. Using a bottom-mounted L-shaped array of hydrophones, the spatial distribution, frequency, and temporal characteristics of sounds produced by small-scale biological processes were estimated within a 40 m by 40 m region around the array. A fast cross-correlation guidance technique lessened the computational burden imposed by conventional and white noise constrained adaptive focusing (curved-wavefront beamforming). Likely sources of biological sound and variation of the acoustic field within and between the sample sites will be discussed. The variation of sound field properties over latitude and correlation with ecological information obtained at each site may be useful in underwater biological surveys that utilize passive acoustic recording.

Session 2pEA

Engineering Acoustics: Transduction

Dehua Huang, Chair
NUWC, 43 Holliston Ave., Portsmouth, RI 02871

Contributed Papers

1:30

2pEA1. Excitation of multi-modes of vibration using tangentially polarized thin walled cylinders. Sairajan Sarangapani (Rowe Technologies, Inc., 12136, Via Milano, San Diego, CA 92128, ssairajan@yahoo.com) and David A. Brown (Electro-Acoust. Res. Lab., Adv. Technol. and Manufacturing Ctr. and ECE Dept., BTech Acoust. LLC, Univ. of Massachusetts Dartmouth, Massachusetts, MA)

Tangentially polarized thin-walled striped-electroded piezoelectric cylindrical transducers are used in several electromechanical and electroacoustic applications. This study is an extension of the previous work [J. Acoust. Soc. Am. **133**, 2661 (2013); J. Acoust. Soc. Am. **132**, 3068 (2012)] and involves the study of electromechanical excitation of multi-modes of vibration using tangentially polarized cylinders. A numerical finite difference method (FDM) was used to analysis the nonuniform electric field in the tangentially polarized cylinder under the assumption that the piezoelectric element is fully polarized. The electromechanical properties including the effective electromechanical coupling coefficient, effective piezoelectric modulus, the effective compliance, and effective relative dielectric constant were calculated for a tangentially polarized cylinder for several modes of vibration and compared with results of cylinders using traditional transverse piezoelectric effect.

1:45

2pEA2. Statistical properties of random arrays. Jenny Au, Charles Thompson, and Ololade Mudasiru (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, Jenny_Au@student.uml.edu)

In this work, the statistical properties of the randomly placed linear arrays is considered. The performance of random, random-bin and nearest-neighbor constraint placement of transducer elements is considered. Transducer placement locations drawn from the transducer density function satisfying the van der Maas objective function is of particular interest. It is shown that the variance in the response of the sidelobe region decays asymptotically as 1/N, where N is the number of transducers.

2:00

2pEA3. Construction of a new underwater low-frequency projector based on clarinet acoustics. Andrew A. Acquaviva (Acoust., Penn State Univ., 871 Willard St., State College, PA 16803, acquavaa@gmail.com)

Understanding theoretical models of novel designs is a crucial component of developing early prototypes. This presentation discusses a new underwater low-frequency sound source. This system is based on a model of a clarinet-like source that is designed to work underwater, such that the complete assembly incorporates all of the necessary components required for a real clarinet to produce sound. The resulting device can, with proper set up and understanding of clarinet acoustics, produce a stable tone at significant amplitudes. Furthermore, the harmonic content substantiates the validity of the model on which it is based.

2:15

2pEA4. Design of piezo-micro-electro-mechanical systems for low frequency energy harvesting. Swapnil D. Shamkuwar and Kunal N. Dekate (Electronics, G.H.Raisoni College of Eng.,Nagpur, 96 Naik Nagar, Post Parvati Nagar, Nagpur, Maharashtra 440027, India, shamkuwarswapnil@gmail.com)

The exibility associated with piezoelectric materials makes them very attractive for power harvesting. Piezoelectric materials possess a large amount of mechanical energy that can be converted into electrical energy, and they can withstand large strain magnitude. The critical physical dimensions of MEMS devices can vary from well below one micron on the lower end of the dimensional spectrum, all the way to several millimeters. While the functional elements of MEMS are miniaturized structures, sensors, actuators, and microelectronics, the most notable (and perhaps most interesting) elements are the microsensors and microactuators. Microsensors and microactuators are appropriately categorized as "transducers," which are defined as devices that convert energy from one form to another. In the case of microsensors, the device typically converts a measured mechanical signal into an electrical signal. Mechanical compression or tension on a poled piezoelectric ceramic element changes the dipole moment, creating a voltage. Compression along the direction of polarization, or tension perpendicular to the direction of polarization, generates voltage of the same polarity as the polling voltage. Hence, by changing device physics, we may get a sensor with higher output with low power consumption and reduced size of the device. The proposed piezoelectric sensor will be designed in COMSOL software and respective characteristics analysis will be observed.

2:30

2pEA5. Ultrasound communication for body senor network. Meina Li, Canghee Hyoung, Junghwan Hwang, Sungweon Kang, and Kyunghwan Park (Human Interface SoC, Electronics and Telecommunications Res. Inst., 218 Gajeongno, Yuseong-gu, Daejeon 305700, South Korea, limeinajl85@etri.re.kr)

Radio frequency (RF) waves have been the dominant part in the field of wireless communication. However, RF waves has the limitation when transmit through the human due to the most of body composition is water. Therefore, the ultrasound that has good propagation in the water has been proposed as the new communication wave for the body sensor network (BSN). Two ultrasonic sensors are used in the communication system. One was placed under skin as the transmitter (Tx) and the other one as the receiver (Rx) was placed right toward the Tx on the top of skin. The Tx can detect and monitor the physiological signal inside of body then transmit the information to the Rx. After the Rx received the signal, it can transmit the information to the doctor also can release the drug to the patient. The modulation of ultrasonic wave has been experimented by three common modulation ASK, PSK, FSK for digital communication. The results showed the ultrasound wave is a possible communication method for body sensor network.

2:45

2pEA6. An experimental and computational study of beam-steering of parametric array. Kyunghun Been, Yub Je, and Wonkyu Moon (Mech. Eng., Pohang Univ. of Sci. and Technol., POSTECH, San 31, Hyojadong, Namgu, Pohang 790-784, South Korea, khbeen@postech.ac.kr)

A parametric array is a nonlinear conversion process that can generate a highly directional sound beam with a small aperture. It is expected that electrical beam steering of directional sound beams generated by the parametric array may be useful in many applications such as ultrasonic raging sensors or directional loudspeakers in air. One of the major issues of beam steering of the parametric array is to precisely predict the steered difference frequency wave field in the medium. In this study, beam steering of the parametric array is computed by using a time-domain numerical code that solves the Khokhlov-Zabolotskaya-Kuznetsov equation. Since it is impossible to compute the exact difference wave field due to a complex primary source distribution in the medium, a simplified numerical model is proposed. The computed result is compared with the experimental result. For experimental study, 16-channel piezoelectric micromachined ultrasonic transducer array, which consists of two resonant type unit drivers to generate bi-frequency primary waves ($f1 = 100$ kHz and $f2 = 140$ kHz), was designed, fabricated, and tested. The beam patterns of the primary and difference frequency waves were measured and compared with the computed result while applying complex weighting to each channel. [Work supported by ADD (UD130007DD).]

3:00

2pEA7. A micro-machined hydrophone based on piezoelectric-gate-of-field-effect-transistor for low frequency sounds detection. Min Sung, Kumjae Shin (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology (POSTECH), PIRO 416, Hyoja, Namgu, Pohang City, Kyungbuk, Pohang, Kyungbuk 790784, South Korea, smmath2@postech.ac.kr), Cheeyoung Joh (Underwater Sensor Lab., Agency for Defense Development, Changwon, Kyungnam, South Korea), and Wonkyu Moon (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), Pohang, Kyungbuk, South Korea)

The miniaturization of conventional piezoelectric hydrophone is known to have limits in low frequencies due to high sensor impedance of micro-sized piezoelectric body. In this study, a new transduction mechanism is devised and named as piezoelectric gate of field effect transistor (PiGoFET) so that its application could solve the sensitivity limitation of a miniaturized hydrophone with a tiny piezoelectric body less than 1 mm. The PiGoFET transduction can be realized by combination of a field effect transistor and a small piezoelectric body on its gate. We connect a micromachined membrane to the small piezoelectric body so that acoustic pressure can apply appropriate forces on the body on the FET gate. The electric field from the strained piezoelectric body modulates the channel current of FET at any frequency less than high limit of transistor; thus, the sound pressure may be transferred to the source-drain currents even at very low frequencies irrespective of the size of piezoelectric body. Under the described concept, a small hydrophone was fabricated by micromachining and calibrated using the comparison method in low frequencies to investigate its performance as a low frequency sensitive hydrophone. [Research funded by MRCnD.]

3:15

2pEA8. A piezoelectric micro-cantilever acoustic vector sensor. Wonkyu Moon, Sungkwan Yang (Dept. of Mech. Eng., Pohang Univ. of Sci. and Technology(POSTECH), PIRO 405, POSTECH, San31, Hyoja-dong, Namgu, Pohang City, Kyungbuk 790784, South Korea, wkmoon@postech.ac.kr), Joh Cheeyoung, and Kyungsub Kim (Underwater Sensor Lab., Agency for Defense Development, Changwon, South Korea)

An acoustic vector sensor measures the direction of wave propagation as well as the acoustic pressure. We investigate feasibility of using a piezoelectric micro-cantilever (PEMC) as an acoustic vector sensor in water at the frequencies range below 500 Hz. In order to measure the propagation direction, we try to devise the properties of a PEMC so that its deflection is proportional to the particle velocity due to acoustic waves. We found that the desired property can be obtained with PEMC if it is designed to be flexible enough. Under the assumption that the PEMC affects little on the wave propagation, we have developed a simple lumped parameter model to predict the relationship between the acoustic pressure of a progressive wave and the deflection of PEMC. The developed model shows that the deflection of PEMC is dependent on the magnitude and direction of the incoming progressive wave. In addition, the frequency of the wave is also found to affect the responses of PEMC. Based on the developed simple lumped parameter model, a PEMC acoustic vector sensor was designed with PEMC of 400 μm long 200 μm wide 5.25 μm thick for operating around 200 Hz. The designed PEMC acoustic vector sensor was fabricated by micro-machining and packaged inside cater-oil-filled rubber housing so that it can be tested in the water. The expected dependence of the fabricated PEMC on the direction and acoustic pressure can be observed in the experiments at the target frequency range.

Session 2pMU

Musical Acoustics: Acoustics of the Organ

Uwe J. Hansen, Chair
Indiana State Univ., 64 Heritage Dr., Terre Haute, IN 47803-2374

Chair's Introduction—1:00

Invited Papers

1:05

2pMU1. Sound choices: Designing a pipe organ to suit its acoustical environment. Matthew M. Bellocchio (Andover Organ Co., P. O. Box 36, Methuen, MA 01832, mbellocchio@andoverorgan.com)

A key to the effectiveness of any pipe organ is to design it to be acoustically compatible with its environment. Churches and concert halls all vary in size, plan, seating arrangement, resonance, reflectivity, and reverberation. A successful instrument should have a balanced sound in the room throughout its frequency range, be capable of both warmth and brilliance, and have a variety of tonal resources suitable for both solo performance and accompaniment. The choices an organbuilder makes regarding placement, physical size and layout, tonal specification (choice of stops), pipe materials and dimensions, wind pressures, and voicing style, all contribute to an instrument's tonal signature and acoustical effectiveness. These decisions are usually based on historical precedents, traditional knowledge, personal taste, and experience. The four-manual Casavant organ, Opus 3145, in the Cathedral of Saints Peter and Paul in Providence, Rhode Island, illustrates Lawrence Phelps' tonal design for a heroically sized mechanical action instrument intended for a large, acoustically live space. This organ, built in 1972 at the height of the Baroque Revival period in American organbuilding, was intended to be a legacy instrument for the designer, the builder, and the client.

1:25

2pMU2. Room acoustics and pipe organs: Putting a STOP to common misunderstandings. Neil T. Shade (Acoust. Design Collaborative, Ltd., 7509 Lhirondelle Club Rd., Ruxton, MD 21204, nts@akustx.com)

Misunderstandings about room acoustics exist among architects, building committees, musicians, organ consultants, and organ builders. These range from opinions with no acoustic validity to simple confusion of acoustic principles by those not versed in our science. Common and repeated mistakes this author has encountered while serving as an acoustic consultant to numerous worship houses are discussed along with a rational basis to correct such unfounded beliefs.

1:45

2pMU3. A survey of pipe organ reed research emphasizing recent developments. George Plitnik (Phys., Frostburg State Univ., 120 Compton Hall, Frosrburg, MD 21532, gplitnik@frostburg.edu)

As experimental and theoretical acoustics advanced during the 19th century several prominent researchers attempted, typically unsuccessfully, to understand the physics of pipe organ reeds. Centuries of organ building experience had advanced the art of reed voicing to a consummate skill passed down from masters to apprentices, with no understanding of, and no desire to learn, the underlying physical principles. This presentation will give a cursory survey of two centuries of organ reed investigations and then highlight the research during the past two decades which has unlocked some of the conundrums of the past to render these complicated instruments amenable to scientific understanding. The following reed parameters and their influence tone will be examined: reed thickness, reed length, imposed air pressure, reed curvature, shallot type, shallot filling, and the type of metal used for the reed as well as the provenance and method of its manufacture.

2:05

2pMU4. The influence of the shallot shape on the sound of Trompete reed pipes. Judit Angster, Kai Dolde (Acoust., Fraunhofer IBP, Nobel Str. 12., Stuttgart, 70569, Germany, Judit.Angster@ibp.fraunhofer.de), Rucz Peter (3Budapest Univ. of Technol. , Budapest, Hungary), and Andras Miklos (Steinbeis Transfer Ctr. of Appl. Acoust., Stuttgart, Germany)

The focus of investigations was the influence of certain shallot parameters on the sound of reed pipes. Since the dimensioning of organ pipes is mainly based on the experience of organ builders at present, the aim is to provide scientific data for organ builders explaining the influence of various shallot parameters on the sound. Due to this knowledge, it would be possible to realize the desired ideas of sound of organ pipes by targeted adjustment of these parameters during the dimensioning of the pipes. The investigated Trompete shallots differ in the termination angle of the shallot. To analyze the parameter of the shallot termination, angle measurements were carried out at a simple shallot model, which provide findings on the dependence of the reflection properties on various termination angles. The results show that the findings due to the model comply with the analyses of the sound spectra of the Trompete shallot.

2:25–2:40 Break

2:40

2pMU5. Effect of generators and resonators on musical timbre in coupled systems. Jonas Braasch (Ctr. for Cognition, Commun., and Culture, Rensselaer Polytechnic Inst., 110 8th, Troy, NY 12180, braasj@rpi.edu)

The majority of Organ pipes are typical tone-generator/resonator systems, where the tone is either produced by an air jet or reed. My previous work on free reeds in pipe organs has raised my interest in how different sound generators affect timbre in terms of spectral balance and attack phase. Most recently, I have started to explore different tone generators with the soprano saxophone using custom adapters, including a brass mouthpiece from a cornetto, a bassoon reed and a free reed from the Chinese Bawu. Since the resonator remains the same, the role of the sound generator can be easily determined using a fast Fourier transformation and onset analysis of partial tones. With the brass mouthpiece, the soprano saxophone becomes an outward striking mechanism instead of a reed-based, inward striking mechanism. As a consequence, the instrument vibrates above the natural frequency of the conical resonator, and the B-flat instrument becomes a B instrument. It is important not to underestimate the range of the embouchure. In the case of the cornetto mouthpiece, it took some practice to attain the higher harmonics. Naturally, the spectrum with the cornetto mouthpiece is very similar to that of a regular soprano saxophone. [Work supported by NSF #1002851.]

3:00

2pMU6. Sound production in the reed organ and harmonium. James P. Cottingham (Phys., Coe College, 1220 First Ave., Cedar Rapids, IA 52402, jcotting@coe.edu)

The free reed instruments of European origin, which originated around 1800 and were developed over the next 50–75 years, include the reed organ and the harmonium. These keyboard instruments enjoyed a period of great popularity beginning around 1840 and lasting until the early 20th century. They were widely used in homes and churches, and appeared in a variety of instrumental ensembles, including the salon orchestra. They are close relatives of the harmonica and the accordion-concertina family, but unlike these instruments they are not in wide use today. This paper discusses the fundamental mechanisms of sound production in the instruments as well as means used to alter the tone quality, which include the design of the wind system, effects of the chambers in which the reeds are mounted, and details of the reed tongue design. The presentation will include audio examples of the reed organ and the harmonium employed in various musical contexts.

3:20

2pMU7. The Musical Instrument Digital Interface (MIDI): The digital organ for organists and non-organists. Paul Wheeler (Elec. & Comput. Eng., Utah State Univ., 1595 N. 1600 E., Logan, UT 84341, paul.wheeler@usu.edu)

The Musical Instrument Digital Interface (MIDI) is a technical protocol (standardized in 1983) to connect a wide variety of electronic musical instruments. Digital organs, due to increased sound quality and lower pricing, have become a popular alternative to pipe organs. They also provide an advantage of controllability through MIDI. Using MIDI, digital organs can be connected to a variety of sound modules, greatly increasing the number of stops available for organists. For the non-organist composer, organ music can be written in notation software (such as Finale) and played on an actual digital organ. The challenge in writing music for playback through MIDI is to incorporate organ techniques (such as shortening of repeated notes, legato playing by finger crossing or finger substitution, and finger glissandos) so that the result does not sound like a pianist (or a computer) playing the organ. A MIDI capable digital organ can be advantageous for organists and non-organists alike.

3:40

2pMU8. Bach with sampled sounds. Uwe J. Hansen (Chemistry& Phys., Indiana State Univ., 64 Heritage Dr., Terre Haute, IN 47803-2374, uwe.hansen@indstate.edu) and Norman C. Pickering (Pickering Res., East Hampton, NY)

A number of years ago Norman used a digital recorder to record sound samples from a Casavant organ with two manuals plus pedals, with a total of 13 stops in 16 ranks of 882 pipes. He did this by recording all pipes for every other key. He included the initial transient for each sample. Additional organ stops were synthesized from multiple oscillators of various waveforms, based on FFT analysis of actual organ stops. All these sounds were mapped onto the appropriate keys of two synthesizer keyboards. Using these sampled sounds, Norman recorded nearly the entire Bach Orgelbuechlein and several other major works including six Schuebler Chorales, a concerto, BWV 594, and a Prelude and Fugue, BWV 552. We will listen to a brief excerpt from "Wachet auf" to hear the effect of the initial transients. A portion of the Prelude of BWV 552 will give an impression of the full organ.

Session 2pNS

Noise and Architectural Acoustics: Acoustics During Construction

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

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

Chair's Introduction—1:00

Invited Papers

1:05

2pNS1. The integration of acoustics in construction management education. Norman H. Philipp (School of Construction, Pittsburg State Univ., 1840 E. 153rd Circle, Olathe, KS 66062, nphilipp@geileracoustics.com)

An overview of the importance and need for acoustics and proper noise control mitigation in assemblies and during construction being integrated into construction management education. Items discussed include the impact of noise on a construction site, workers, and neighboring areas; importance of considering acoustical considerations in project scheduling; and methodologies for enabling student understanding of acoustics in construction.

1:25

2pNS2. Mitigation of construction noise at operating hospital and university facilities. Kerrie G. Standlee (Daly-Standlee & Assoc., Inc., 4900 SW Griffith Dr., Ste. 205, Beaverton, OR 97005, kstandlee@acoustechgroup.com)

Hospitals and universities often have occasions where new or remodeling-related construction must occur during times when existing facilities are in use. Noise caused by construction at an active facility often requires that steps be taken to minimize the impacts associated with the construction. This paper discusses the findings made during the course of several projects undertaken by Daly-Standlee & Associates, Inc., to determine mitigation measures that could be used to minimize noise impacts from construction at operating hospital and university facilities.

1:45

2pNS3. Noise control for an impact pile driver in an urban environment. Jerry G. Lilly (JGL Acoust., Inc., 5266 NW Village Park Dr., Issaquah, WA 98027, jerry@jglacoustics.com)

Soil conditions at a construction site for a new seven-story office building in downtown Seattle, Washington, dictated the use of an impact pile driver to set 50 to 60-ft long concrete piles into the ground, many less than 200 ft from existing offices. Exterior and interior noise measurements revealed that the pile driving activity exceeded the city's noise ordinance. A portable acoustic shield was designed, constructed, and implemented to reduce noise levels below the legal limits. Measured sound levels before and after will be presented along with details and photos of the shield in action.

2:05

2pNS4. Construction issues affecting the intended acoustical environment of a new court tower. Robert M. Brenneman (Acoust., McKay Conant Hoover, Inc., 7435 E Stetson Dr., Ste. A, Scottsdale, AZ 85251, rbrenneman@mchinc.com)

The proficiency to readily identify and resolve construction issues in the field is vital to realizing intended acoustical design goals. In court facilities, inattention to such construction details can result in problematic reductions in speech privacy for confidential deliberations, negotiations, and attorney discussions, structure-borne noise intrusion from holding areas to noise-sensitive spaces, and diminished speech intelligibility in courtroom proceedings. This paper considers examples of acoustics and noise control issues arising during the construction of a new court tower, discussion of their potential effects on the acoustical environment, and methods of addressing these field conditions.

2:25

2pNS5. Preparation, execution, and documentation of construction site visits. Neil T. Shade (Acoust. Design Collaborative, Ltd., 7509 Lhirondelle Club Rd., Ruxton, MD 21204, nts@akustx.com)

Construction site visits, referred to as "work observation" by the American Institute of Architects, are a critical factor in project delivery. The unique nature of acoustic detailing requires visual and sometimes field testing to verify correct installation of products and construction assemblies. Preparation for site visits begins during the Construction Documents design phase. A list of site inspections, keyed to project milestones, should be issued to the Contractor at the start of the Construction Administration phase. Site visit objectives should be planned and the Contractor notified prior to arriving on site. When on site it is necessary to conduct inspections in a methodical manner, document potential deviations from design intent, and debrief the Contractor. Checklists can improve efficiency and avoid overlooking key objectives. Work observation reports become part of the project record and are important for tracking corrective measures by the Contractor. Acoustic measurements may be part of inspections to check quality of work or determine compliance with Specifications. Well-planned and executed site visits contribute to project success and boost our profession's esteem with the building construction industry.

2:45

2pNS6. A potpourri of acoustical issues arising during construction administration. Joseph F. Bridger and Noral D. Stewart (Stewart Acoust. Consultants, 7330 Chapel Hill Rd.,Ste. 101, Raleigh, NC 27607, joe@sacnc.com)

This is a collection of our firm's most noteworthy experiences during construction administration. First, they illustrate the many ways what is built can go astray from what was recommended. In other cases, we do not get involved with the particular area of scope until construction administration. Sometimes in such cases, design professionals attempt to do acoustics themselves but lack the knowledge to do so correctly. At other times, design professionals are not even aware that an acoustics issue should be addressed, until it is noticed during construction. Of course, there is the contractor whom did not understand or simply did not do what the drawings and specifications called for. Lastly, there can simply be things that cannot/should not be built the way they were designed (often the contractor notices these things first) and the inevitable "now what do we do?" Of course anything that comes up during construction is on a short and critical time schedule, and fraught with the challenge of how to get things fixed quickly and with least added cost. It is our hope that these case studies will help building professionals and acoustical consultants on future projects.

3:05–3:20 Break

3:20

2pNS7. On the importance of experienced acoustical inspectors during and following the construction process and installation of noise mitigation measures. Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

During the installation of recommended noise mitigation measures that appear on architectural drawings or that are presented in project acoustical reports, demolition, renovation, office building, institutional facilities, apartment buildings, and condominium construction trades may or may not omit and/or improperly install the recommended noise control systems. This paper presents several examples of such improper practices that, had they not been discovered, would have resulted in significant reductions in the planned noise control performance. These cases result in undesired living and/or working conditions. The selected examples of such construction problems are discussed and successful remedies are presented, some of which avoided threatened litigation.

3:40

2pNS8. Issues and opportunities during construction of a new courthouse building. K. Anthony Hoover (McKay Conant Hoover, 5655 Lindero Canyon Rd., Ste. 325, Westlake Village, CA 91362, thoover@mchinc.com)

A new courthouse building was recently completed, housing 31 courtrooms and the full complement of ancillary spaces, and using a delivery system called performance-based infrastructure (PBI) that enhanced the need for high durability and long-term functionality, as well as a fast-paced schedule. An interesting issue arose concerning unexpectedly loud levels of toilet flush noise in a public restroom adjacent to a judge's chamber, whose resolution resulted in 20 dBA reduction at the source without need for reconstructed partitions in over a dozen similar adjacencies. An interesting opportunity arose when on opening day it was determined that a cueing room was noisy, and one of the architects emailed audio files of balloon pops recorded on his cell phone, in a manner that simulated measurements that he had witnessed several times before, and which confirmed our calculations sufficiently to engender agreement on treatment without need for additional site visits. These and perhaps other issues and opportunities will be discussed.

4:00

2pNS9. Direct and flanking airborne sound transmission for hotel guestrooms; results and comments regarding three field tests. Jim X. Borzym (Borzym Acoustics LLC, 2221 Columbine Ave., Boulder, CO 80302, acoustics@columbine.netr)

A "four-and-a-half star" hotel was constructed within an existing historic structure. Among the design/build team there were differences of opinion of what was needed for peace and privacy for hotel guests. Hotel guidelines suggested a moderately high airborne sound transmission class rating for the primary wall partition. Three field tests were conducted during an early phase of construction. A strong flanking transmission path in a corridor wall assembly was discovered during the first test. An error in preparation for this test caused an important opportunity for valuable field test data development to be lost regarding a flanking path via a common ceiling, which points to the kinds of difficulties encountered in conducting field tests. The second test showed improvement in results due to rectification of the corridor wall flanking path. The third test gave information about flanking transmission via the common floor. Comments will be made regarding field testing and flanking sound transmission.

4:20

2pNS10. Construction monitoring—A lack of standardization. Tyler Rynberg (VACC, Inc., 490 Post St, Ste. 1427, San Francisco, CA 94102, tyler@va-consult.com)

Construction noise and vibration monitoring is being required more and more frequently, especially for large infrastructure projects. Work on these civil projects often involves tunneling, piling, large-scale earthworks, and even blasting—all in close proximity to commercial or residential population centers. Aside from the human-related annoyance effects, there are effects on the environment (endangered species) and the possibility of damage to structures. Because construction monitoring for noise and vibration is a relatively young field, there is little guidance for agencies to consult; the result is a multitude of differing monitoring requirements on these projects. From project to project, the monitoring requirements differ in subtle ways that do not necessarily correspond to the true needs of different projects. This presents serious problems in terms of developing consistent protocols (allowing apples to apples comparisons) and also in instrumentation, as most instrument vendors are too inflexible to meet the range of requirements or are too expensive for the flexibility they do offer (or both). A discussion to develop more consistent standards in construction monitoring is needed.

Contributed Papers

4:40

2pNS11. Method to improve warning sound detection for hearing protectors. Eric Bernstein, Anthony J. Brammer, and Gongqiang Yu (Univ. of Connecticut Health Ctr., 263 Farmington Ave., Farmington, CT 06030, eric.r.bernstein@gmail.com)

Hearing protection devices (HPDs) provide both desirable attenuation of environmental noise and undesirable attenuation of auditory warning alarms, such as vehicle backup alarms. Both warning alarm detection and localization performance are affected. A digital cross-correlation based method is described to identify a pre-selected warning alarm to bypass the attenuation of the HPD while maintaining attenuation of environmental noise outside the bandwidth of the alarm. This method can be integrated into existing digital HPD designs. Computer simulation of the algorithm demonstrates that an alarm signal can be detected at signal-to-environmental noise ratios as low as 30 dB for the military and industrial noise sources investigated. Implementation of the method using a modified commercial HPD demonstrates a 7 dB improvement in warning alarm detection threshold compared with an unmodified HPD. Alternative methods for presenting the alarm signal to the user will be discussed as well as modifications to expand the method to accommodate multiple alarm signals.

4:55

2pNS12. Reduction of construction noise while building a recording studio. Sergio Beristain (IMA, ESIME, IPN, P.O.Box 12-1022, Narvarte, Mexico City 03001, Mexico, sberista@hotmail.com)

While a new recording studio was being built within a large Audio and Video complex, construction noise had to be adequately controlled, in order to allow for the already existent audio recording studios and TV sets to continue with their normal activities along the building process. The noise was reduced by making a double wall system in the façade overlooking the construction area. Use was made of the already existing external wall, adjacent to the construction site, as the first wall of the system, and a second one was added in the exterior side of that wall in order to complete the double wall system, which contributed with a few dB's in the low frequency range, while getting a considerable noise level reduction in the high frequencies range to the overall noise reduction of the original wall. Producers and performers who were actually working in the existing studios were satisfied by the results obtained once the adaptation was complete. Details of the wall and its expected acoustical behavior are presented and discussed.

Session 2pPA

Physical Acoustics and Education in Acoustics: Demonstrations in Acoustics

Murray S. Korman, Chair
Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402

Chair's Introduction—12:55

Invited Papers

1:00

2pPA1. Apparatus for demonstrating evanescent waves in acoustic waveguides. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu) and Daniel O. Ludwigsen (Dept. of Phys., Kettering Univ., Flint, MI)

A physical demonstration apparatus, inspired by [K. Meykens, *et. al*., Am. J. Phys., **67**(5), 400–406 (1999)] is used to demonstrate evanescent sound waves in a rectangular waveguide. The apparatus consists of a rectangular waveguide, approximately one meter in length, with one optically transparent wall. The waveguide is driven at one end by a pair of loudspeakers whose polarity may be switched. The other end of the waveguide is open. The sound field inside the waveguide may be interrogated using a microphone and an oscilloscope. The apparatus will be used to demonstrate several features of acoustic wave guides, including the propagation of plane waves, the cutoff frequency for non-plane waves, the propagation of non-plane waves above the cutoff frequency, and the exponential decay of evanescent non-plane waves below the cutoff frequency. The educational and teaching applications of this apparatus will be discussed.

1:20

2pPA2. Fun with foggers. R. Glynn Holt (Dept. of Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, rgholt@bu.edu)

Many physical acoustics phenomena can be demonstrated with a so-called "pond fogger." The easy availability and affordable price of these acoustic transducer systems allows one a lot of freedom in the demonstrations one can do. Three phenomena, acoustic streaming, standing waves, and Faraday waves, will be demonstrated in this talk. The audience will be invited to propose and investigate others during the talk.

1:40

2pPA3. Use of the hammered dulcimer to demonstrate physical acoustics principles. Cameron T. Vongsawad, Kent L. Gee, Tracianne B. Neilsen, and Benjamin Y. Christiansen (Dept. of Phys. & Astronomy, Brigham Young Univ., 1041 E. Briar Ave., Provo, UT 84604, cvongsawad@byu.net)

In 1636 AD, Marin Mersenne described the law of vibrating strings that relates frequency to length, tension, and density in *L'Harmonie Universelle*. One of the instruments described by Mersenne is the *psalterion* or hammered dulcimer. The dulcimer is a versatile struck-string instrument, based on the circle of fifths, in which the sound generation and radiation are linked to many physical acoustics principles. Some of these principles are basic: how string properties affect fundamental frequency; how the soundboard and body cavity give rise to different resonances; how hammer construction and excitation affect the spectrum; etc. Because of the ease of demonstrating acoustical principles with the hammered dulcimer, a 9/8 backpack-size dulcimer has been included as part of the new ASA outreach workshops held in conjunction with semiannual meetings. A description of these efforts and a short demonstration of the hammered dulcimer will be given. More advanced concepts to be discussed include high-speed video results of the hammer-string interaction as well as near-field acoustical holography analyses on a 16/15 Songbird® dulcimer. The holography shows how the sound holes, traditionally thought to be merely decorative, significantly influence the radiated sound.

2:00

2pPA4. Demonstration of an extremely directional acoustic source. Preston S. Wilson, Craig N. Dolder, and Mark F. Hamilton (Mech. Eng. and Appl. Res. Labs., The Univ. of Texas at Austin, 1 University Station, C2200, Austin, TX 78712, pswilson@mail.utexas.edu)

Highly directional light sources such as flashlights and lasers are well known to most people. In contrast, highly directional acoustic sources, or in other words, sources of sound that are audible in only a very narrow region of space, are far less common. Many people have never experienced such a source, and the phenomenon is not found in nature. A highly directional source of sound known as a parametric array is used underwater for sonar applications, but the frequency (pitch) of the sound is often above the human hearing range. Similarly, highly directional, focused sound sources are regularly used in medical applications, but again, the frequency is too high to be

heard. The narrowness of the acoustic beam cannot be experienced by human listeners. Recently, parametric array technology has been commercialized for use in air at frequencies in the human auditory range. These devices produce extremely narrow (on the order of 2 degrees) beams of audible sound. When pointed directly at one listener, the sound is virtually inaudible to another listener only a few feet away. Such a device will be demonstrated and the basic physics behind its operation will be explained.

2:20

2pPA5. Acoustics concepts you can demonstrate with a coffee mug. Andrew C. Morrison (Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431, amorrison@jjc.edu)

The simple coffee mug can be used to illustrate many important concepts in acoustics. Adding water to the mug shows the effect of mass on the frequency of oscillation. Striking on the handle of the mug compared to striking the side of the mug shows the vibrational mode degeneracy lifting. Adding instant coffee or hot chocolate mix to a mug full of water heated in a microwave oven shows the effect of bubble density on the speed of sound. Each of these concepts are easily demonstrated and can also be used as starting points for student exploration in the laboratory.

2:40–3:00 Break

3:00

2pPA6. Nonlinear experiments of tuning curve resonances for the vibrational modes of a weakly stretched circular membrane. Benjamin W. Lloyd and Murray S. Korman (Phys. Dept., U.S. Naval Acad., 572 C Holloway Rd., Chauvenet Hall Rm. 295, Annapolis, MD 21402, korman@usna.edu)

Nonlinear oscillations of a pre-stretched 11.5 cm diameter latex circular membrane clamped rigidly around the boundary were investigated. The relaxed membrane thickness was 0.4 mm. An 8 cm diameter loud speaker drove the membrane from below using a swept sinusoidal tone. Low drive tuning curve sweeps demonstrated resonant frequencies corresponded to the first three radially symmetric linear drum modes (56, 147.6, and 240.5 Hz) sweeping from 50 to 300 Hz. Regression yielded a transverse wave speed of 10.6 m/s. Incrementally increasing the speaker amplitude (after each sweep) showed the lowest resonant frequency at first decreases, then increases with increasing acoustic drive pressure. Higher modes exhibited frequency increases with drive level. Tuning curve responses were measured using a 6 mm diameter microphone 5 mm above the center of the membrane. Nonlinear tuning curves hysteresis effects occur for higher drive levels sweeping between 50 and 70 Hz. Here, the amplitude sharply falls from a high to a low microphone response at f_a = 59 Hz. With the same drive level, a sweep tone starting from 70 Hz to 50 Hz exhibits at f_a = 57.5 Hz, a jump from a lower to a higher microphone response. [Goncalves *et al*., J. Vib. Sound **327**, 231 (2009).]

3:20

2pPA7. Non-electric sensing, amplification, and presentation of sound using fluidic laminar flow technology. Michael V. Scanlon (US Army Res. Lab., RDRL-SES-P, 2800 Powder Mill Rd., Adelphi, MD 20783-1197, michael.v.scanlon2.civ@mail.mil)

This briefing will describe and demonstrate fluidic-based sound reception, amplification, and pneumatic headphone presentation to the listener. Fluidics is non-electronic and uses laminar proportional amplifiers and fluid-flow technology to demonstrate a sensitive "microphone" due to its massless diaphragm (laminar jet of low-velocity air) and perfect impedance matching to the propagating medium (also air). Many years ago the U.S. Army developed a system called the Individual Soldier Operated Personal Acoustic Detection System (ISOPADS) that utilized this fluidic technology to enhance and extend the Soldier's listening range. This demonstration system will let people point a small parabolic dish and listen around the room using a pneumatic headset. It makes for a fun demo, especially when participants understand that they are listening to a non-electronic system that uses only air.

3:40

2pPA8. Generation of photoacoustic transients from optically induced thermal gradients. Clifford Frez, Binbin Wu, and Gerald Diebold (Dept. of Chemistry, Brown Univ., Box H, Providence, RI 02912, Gerald_Diebold@Brown.edu)

Irradiation of an absorbing surface in contact with a transparent fluid with a pulsed laser can result in the generation of extremely large thermal gradients. For example, when a laser with a pulse width of 10 ns and a fluence of 1 J/cm^2 irradiates a region with an absorption of 1 cm^1 having the thermal properties of liquid water, a thermal gradient on the order of 105 K/m at the interface is produced. Here, it is shown that the effect of such thermal gradients on photoacoustic waves from an infinite half space and from a uniformly irradiated sphere is the production of fast transients on the leading edges of the waves. The character of the transients is determined from an additional source term in the wave equation for pressure that obtains when heat conduction is taken into account. Experiments are reported showing the predicted transients on photoacoustic waves from absorbing layers in contact with transparent fluids irradiated with 10 ns laser pulses.

4:00

2pPA9. Investigation of capillary wave formation on water jets with internally propagating ultrasound. Nikhil M. Banda (ISVR, Faculty of Eng. and Environment, Univ. of Southampton, Southampton, Hampshire SO17 1BJ, United Kingdom, mnb1g10@soton.ac.uk), Offin Douglas, Birkin R. Peter (Dept. of Chemistry, Univ. of Southampton, Southampton, United Kingdom), and Leighton G. Timothy (ISVR, Faculty of Eng. and Environment, Univ. of Southampton, Southampton, United Kingdom)

The formation and control of capillary waves on jets are of importance in applications such as ink-jet printing, atomization of fuel, etc. External control of jet breakup processes is generally based on the ultrasonic atomization principle (used on jets of diameter in micrometer order) with transducers placed on nozzle tips. The present work investigates the formation of capillary waves and subsequent jet breakup on 10 mm and 15 mm diameter water jets with ultrasound propagating at either 121 kHz or 135 kHz. Experimental

observations of the jet breakup process with a high speed camera are reported. The input signal to the transducer was controlled to investigate the formation and growth of capillary waves, leading to the breakup of the jet. It was observed that once the waves are formed on jet surface, they grow in size leading to a necking zone. Once necking zone is formed, the capillary waves then just propagate along the jet (with the flow) with no further growth in their amplitude. Spraying of the jet was also observed at the same time. The measurement of capillary wavelength and jet breakup length are measured and presented in an attempt to understand the nature of the breakup process.

4:20

2pPA10. Comparison of experimental and theoretical mode studies on a square plate. Uwe J. Hansen (Indiana State Univ., 64 Heritage Dr., Terre Haute, IN 47803-2374, uwe.hansen@indstate.edu)

Standard two-dimensional Chladni patterns on a square plate are demonstrated and compared with the results of Finite Element mode calculations

4:40–5:30 Demonstration Interaction

TUESDAY AFTERNOON, 6 MAY 2014 BALLROOM B, 1:25 P.M. TO 5:00 P.M.

Session 2pPP

Psychological and Physiological Acoustics: Scientific Catalyst, Collaborator, and Gadfly: Honoring the Contributions of Tino (Constantine) Trahiotis to the Understanding of Binaural Auditory Processing

Leslie R. Bernstein, Cochair
Neurosci. and Surgery, Univ. of Connecticut Health Ctr., MC3401, 263 Farmington Ave., Farmington, CT 06030

H. Steven Colburn, Cochair
Hearing Res. Ctr. and Biomedical Eng., Boston Univ., 44 Cummington Mall, Boston, MA 02215

Richard M. Stern, Cochair
Electrical and Computer Eng., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213

Chair's Introduction—1:25

Invited Papers

1:30

2pPP1. Constantine Trahiotis and hearing science: A half-century of contributions and collaborations. Leslie R. Bernstein (Neurosci. and Surgery, Univ. of Connecticut Health Ctr., MC3401, 263 Farmington Ave., Farmington, CT 06030, Les@neuron.uchc.edu)

Although best known for his contributions to human binaural processing, Constantine (Tino) Trahiotis has, throughout his career, been an avid student of virtually all aspects of hearing. His published work spans nearly 50 years and has been carried out with an impressive cadre of collaborators worldwide (all of whom he thinks of as family). That body of work attests to the diversity of Tino's interests and encompasses behavioral and lesion studies in animals, human psychoacoustics, mathematical modeling, methodology, signal-processing, and instrumentation. Tino's recall of the literature is legendary. As compared to the use of traditional methods to locate publications concerning a particular topic, many of us, especially the string of scientists he has mentored, know that it is often more efficient to just pick up the phone and ask Tino. This presentation traces the history of Tino's career, including his days as a graduate student with Don Elliott at Wayne State University, his time at Indiana University, and as a Professor at the University of Illinois and at the University of Connecticut Health Center.

1:50

2pPP2. Tino Trahiotis's impact on general binaural research progress. H. Steven Colburn (Hearing Res. Ctr. and Biomedical Eng. Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215, colburn@bu.edu)

The evolution of thinking and understanding in binaural hearing research over the past six or seven decades will be reviewed. Both physiological and psychophysical developments, as well as their interactions and applications, will be described. The focus of this review will be the impactful role that Tino Trahiotis played across this wide spectrum of research areas. Tino's large impact came

through his personal engagement with the ideas and the people involved in this research. His enthusiasm for the topic is contagious and provides fuel for digging deeper and making connections across diverse research topics. The progress has been exciting and the whole process has been great fun... consistent with the Trahiotis style. [Work supported by NIH/NIDCD DC00100.]

2:10

2pPP3. Time and intensity with Tino. Richard M. Stern (Elec. and Comput. Eng., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, rms@cs.cmu.edu)

Over a period of decades Constantine (Tino) Trahiotis has been a contemplative and creative scholar, an excellent teacher, a helpful collaborator, as well as a loyal friend. Tino, along with Bob Bilger and Erv Hafter, provided this author's introduction to the broader science of hearing beyond the particular perspective of my graduate training. At the same time, my attempts to interpret models of binaural interaction with Tino led to the review chapters on binaural modeling that the two of us jointly authored in the 1990s. Tino's demonstration of the unexpected dependence on bandwidth of the laterality of bands of noise presented with interaural temporal differences of large magnitude motivated me to think about the tradeoff between "straightness" and "centrality," which appears to be helpful in considering the lateralization of many binaural stimuli. Finally, Tino, along with Les Bernstein, insisted on a local implementation of my models in their laboratory which became one source (along with significant others) for the binaural modeling toolbox developed by Michael Akeroyd that is presently in wide circulation. This talk will review these facets of my life with Tino, and comment on my current perspectives on related issues. [Work supported by DARPA and Cisco Research.]

2:30

2pPP4. The temporal acuity for processing interaural cues. Steven van de Par and Darrin Reed (Acoust., Univ. of Oldenburg, Carl-von-Ossietzky-strasse 9-11, Oldenburg 26129, Germany, steven.van.de.par@uni-oldenburg.de)

It was during a three month visit of the first author to his Lab in 1998 that Tino pointed out an elegant study on detecting changes in Interaural Cross Correlation of stimuli that were roved in baseline Interaural Time Delay and Interaural Level Difference (Bernstein and Trahiotis, J. Acoust. Soc. Am. **102**, 1113). This study showed that, even though ICC can only change due to the presence of dynamically varying ITDs and ILDs, the ICC can be perceived as a separate cue in certain conditions. This articulate notion of three perceptually independent binaural cues, ITD and ILD, relating to perceived laterality, and ICC, related to perceived width, is applied in low bit-rate audio coding where these cues are used to encode the spatial image of stereo sound recordings. The entanglement of these cues on a signal basis raises the question of the time scale at which these three cues are perceptually evaluated. Some recent findings will be discussed that show that the ITD and ICC can be evaluated with a high temporal acuity of less than 10 (ms) in stimuli where these cues alternate periodically, and that these patterns of alternating binaural cues give rise to a modulation percept.

2:50

2pPP5. "The data are the data": Tino and the importance of empirical observation. Michael Akeroyd (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)

I was a postdoctoral scientist in Tino and Les' binaural psychophysics lab from 1999 to 2001. One of my abiding memories from my time then was the motto "the data are the data": in almost every discussion we had on science, Tino would illustrate a point or an idea by reference to a graph of data. Often the graphs were obscure and unjustly ignored by later scientists—but Tino, with his encyclopedic memory for data, would know just where to look for it in a issue of JASA or an mimeographed manuscript. It is a lesson I have always remembered: that science is fundamentally based on accurate measurements of phenomena, and, though the explanations and interpretations may change, the data are always the data, and they are not undone or falsified by the passage of time. This talk will illustrate the argument with data—after all, what else?—from Tino's, mine, and others' papers. [Work supported by the Medical Research Council and the Chief Scientist Office, Scotland.]

3:10–3:25 Break

3:25

2pPP6. Different ears. Marcel van der Heijden (Neurosci., Erasmus MC, P.O.Box 2040, Rotterdam 3000 CA, Netherlands, m.vanderheyden@erasmusmc.nl)

Most models of binaural processing assume identical inputs from the two ears to the binaural processing stage. From the high accuracy of binaural processing one may expect slight deviations from perfect symmetry of its inputs to affect performance. Such deviations may be systematic such as the tuning differences postulated by stereausis models. Alternatively, asymmetries may simply result from imperfections in cochlear frequency tuning or from "sloppy wiring" projecting to the binaural cells. I will analyze how different types of interaural asymmetry affect predicted performance in psychoacoustic tasks and discuss physiological evidence for imperfect symmetry of the monaural inputs.

3:45

2pPP7. The influence of pause, attack, and decay duration of the ongoing envelope on the extent of lateralization produced by interaural time differences of high-frequency stimuli. Mathias Dietz, Martin Klein-Hennig, and Volker Hohmann (Abteilung Medizinische Physik and Cluster of Excellence "Hearing4all," Universität Oldenburg, Carl-von-Ossietzky Str. 9-11, Oldenburg 26129, Germany, mathias.dietz@uni-oldenburg.de)

Klein-Hennig *et al.* [J. Acoust. Soc. Am. **129**, 3856 (2011)] investigated the influence of the duration of specific modulation cycle segments within the ongoing envelope waveform on the sensitivity to interaural time differences (ITD). The ITD sensitivity, measured in a two alternative forced choice discrimination task, was reported to increase for increasing pause and decreasing attack segment duration. The study also revealed that "on" and decay durations have little to no influence on the threshold ITD. The current study employed a subset of nine envelope shapes from the previous study and measured the extent of lateralization produced by ongoing ITDs with an acoustic pointing task. Lateralization generally increased monotonically with ITD for the measured values of 0.2, 0.6, 1, and 2 ms. An additional condition measured combined lateralization of a 1 ms ITD and an opposing 5 dB interaural level difference. It was observed that the extent of lateralization increases with increasing pause duration or with decreasing attack duration in line with the threshold ITD data. However, the different influence of attack and decay flank on ITD sensitivity translates into significant differences in the extent of lateralization only for a subgroup of subjects.

4:05

2pPP8. Sound source localization: Clicks and click trains. William Yost, Xuan Zhong, and Anbar Najam (Speech and Hearing Sci., Arizona State Univ., Tempe, AZ 85287, william.yost@asu.edu)

Tino Trahiotis and Les Bernstein have provided valuable information about human listeners' ability to process interaural time differences in the envelopes of high-frequency carrier signals. These data have enriched models of binaural processing. In this paper, we explore sound source localization of click (100 microsecond transients) stimuli in the azimuth plane in the free field. We are especially interested in the sound source localization of click trains as they provide stimuli with robust envelope properties. We measured sound source localization accuracy for tones, single clicks, and click trains–unfiltered and filtered in low- and high-frequency regions. The filtering was performed to implicate the role of interaural time and level differences in sound source localization. These data involving clicks will be compared to recent findings from our laboratory involving sound source localization of broadband and filtered noise bursts as compared to sound source localization of umodulated and amplitude modulated tonal carriers. These data suggest that stimulus bandwidth is a major factor determining sound source localization accuracy, and that amplitude modulation plays a small role, at best, in determining sound source localization accuracy. The data involving click trains will help expand this database. [Research supported by the AFOSR.]

4:25

2pPP9. A neural measure of interaural correlation based on variance in spike count. David McAlpine, Simon Jones, and Torsten Marquardt (Ear Inst., Univ. College London, 332 Gray, London WC1X 8EE, United Kingdom, d.mcalpine@ucl.ac.uk)

Interaural Correlation (IAC) is related to variance in Interaural Time Difference (ITD) and Interaural Level Difference (ILD). While normalized IAC can account for behavioral performance in discrimination tasks, so can models directly employing this variance as a cue. Attempts at identifying a neural correlate of IAC discrimination have focused on changes in mean spike count, typically at the peaks of ITD tuning curves. We propose that IAC discrimination relies on variance in spike rates on the slope of neural tuning curves (for ITD). We developed a physiologically based hemispheric-balance model of IAC, where fluctuations in the ratio of activity between left- and right-brain serve as the detection cue to a reduction in IAC from unity, a ratio that is stimulus power invariant. Adjusting model parameters, we find that two orders of magnitude less activity is required in the variance-based, compared with mean spike-rate based, model, in order to achieve the same performance. This adjustment also revealed a necessary neural time integration of 10 ms, which is comparable with physiological estimates. The model was tested by recording neural responses from the midbrain of anesthetized guinea pigs to noise stimuli of various IAC.

Contributed Paper

4:45

2pPP10. Combination of two consecutive monaural "looks" as a spatial hearing cue. Xuan Zhong and William Yost (Speech and Hearing Sci., Arizona State Univ., 975 S Myrtle Ave., Lattie F Coor Hall 2211, Tempe, AZ 85281, xuan.zhong@asu.edu)

Interaural time, level, and spectral differences are the major cues being used for sound source localization in the horizontal plane. Past studies have shown that human subjects could be trained to use monaural spectral cues to localize the sources of sound in the azimuth plane, but performance is poor. The current study investigates whether the combination of two monaural signals at both ears, one after another in time, could benefit sound source localization accuracy. The purpose is to study the possibility that the human subjects can compare a monaural signal at one ear to the short term memory of a prior monaural signal arriving at the other ear. Subjects were asked to judge the position of a loudspeaker that presented a 250-ms, 40 dBA noise burst with a roving spectral contour in a quarter field. The rms sound source localization error was measured in three conditions: (1) a single monaural signal; (2) two consecutive monaural "looks" at identical signals, with an interval of 3 s and (3) normal binaural hearing. It was found that the "two looks" localization performance was better than that in the case of monaural presentation, but is still inferior to that of the binaural presentation. Similar experiments were carried out over headphones. Lateralization performance was compared to the sound-field localization data. Contributions of level and spectral differences will be discussed. [Research supported by the AFOSR.]

Session 2pSA

Structural Acoustics and Vibration, Physical Acoustics, Engineering Acoustics, and Noise: Acoustic Metamaterials II

Christina J. Naify, Cochair
Acoust., Naval Res. Lab., 4555 Overlook Ave. SW, Bldg. 2, 138G, Washington, DC 20375

Michael R. Haberman, Cochair
Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758

Invited Papers

1:00

2pSA1. Harnessing geometric and material nonlinearities to design tunable phononic crystals. Katia Bertoldi, Pai Wang, Sicong Shan, and Sahab Babaee (Harvard Univ., 29 Oxford St., Cambridge, MA 02138, bertoldi@seas.harvard.edu)

We investigate numerically and experimentally the effects of geometric and material nonlinearities introduced by deformation on the linear dynamic response of two-dimensional phononic crystals. Our results not only show that deformation can be effectively used to tune the band gaps and the directionality of the propagating waves, but also reveal how geometric and material nonlinearities contribute to the tunable response of phononic crystals. Our study provides a better understanding of the tunable response of phononic crystals and opens avenues for the design of systems with optimized properties and enhanced tunability.

1:20

2pSA2. Damping and nonlinearity in elastic metamaterials: Treatment and effects. Romik Khajehtourian, Michael J. Frazier, Clémence Bacquet, and Mahmoud I. Hussein (Aerosp. Eng. Sci., Univ. of Colorado Boulder, ECAE 194, UCB 429, Boulder, CO 80309, mih@colorado.edu)

Locally resonant acoustic/elastic metamaterials have been the focus of extensive research efforts in recent years due to their attractive dynamical characteristics, such as the possibility of exhibiting subwavelength bandgaps. In this work, we present rigorous formulations for the treatment of damping (e.g., viscous/viscoelastic) and nonlinearity (e.g., geometric/material) in the analysis of elastic wave propagation in elastic metamaterials. In the damping case, we use a generalized form of Bloch's theorem to obtain the dispersion and dissipation factors for freely propagating elastic waves. In the nonlinear case, we combine the standard transfer matrix with an exact formulation we have recently developed for finite-strain elastic waves in a homogeneous medium to obtain the band structure of a 1D elastic metamaterial. Our analysis sheds light on the effects of damping and nonlinearity on the dispersive characteristics in the presence of local resonance.

1:40

2pSA3. Dynamics of geometrically reconfigurable one dimensional and two dimensional magneto-elastic metamaterials. Massimo Ruzzene and Marshall Schaeffer (Georgia Inst. of Technol., 270 Ferst Dr., Atlanta, GA 30332, ruzzene@gatech.edu)

Periodic structures are presented as metamaterials that exhibit multistability due to the nonlinearities of magneto-elastic interactions and structure geometry. The multistability of these structures affords them the ability to adapt their properties though geometric reconfiguration, bringing about changes in stiffness and Poisson's ratio, and introducing anisotropy. These changes in structural properties cause drastic changes in wave propagation, which is of interest for wave control. The dynamic transformation of one-dimensional (1D) and two-dimensional (2D) lattices between stable states are studied through nonlinear numerical simulations. The analysis is conducted using a lumped mass system of magnetic particles. The structures studied include hexagonal, re-entrant, and kagome lattices. Changes in plane wave propagation properties are predicted by applying Bloch theorem to lattice unit cells with linearized interactions. Results from Bloch analysis are then verified through direct numerical simulations. The propagation of plane waves in these lattices before and after topological changes is compared, and large differences are evident.

2:00

2pSA4. Microscale granular metamaterials. Nicholas Boechler (Dept. of Mech. Eng., Univ. of Washington, Mech. Eng. Bldg., Box 352600, Seattle, WA 98195, boechler@uw.edu)

Locally resonant metamaterials and granular media are both known to drastically affect acoustic wave propagation. However, there are thus far few examples of such materials which have microscale elements and are designed to control acoustic waves with megahertz frequencies or greater. In this talk, I will discuss our recent work at the intersection of these two types of materials, in which we explored the interaction of megahertz-gigahertz frequency surface acoustic waves with a self-assembled metamaterial composed of microspheres

adhered to an elastic substrate. I will present our theoretical model and describe our photoacoustic experiments, in which we used transient-grating spectroscopy to measure the acoustic dispersion of the system. Finally, I will also discuss several potential applications of these novel materials such as signal processing and biosensing devices.

Contributed Papers

2:20

2pSA5. Nonlinear behavior of heterogeneous materials containing snapping acoustic metamaterial inclusions. Stephanie G. Konarski, Kyle S. Spratt, Michael R. Haberman, and Mark F. Hamilton (Appl. Res. Labs., The Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713-8029, skonarski@utexas.edu)

This work studies the forced dynamical behavior of a heterogeneous material containing metamaterial inclusions undergoing large deformations. The inclusions exhibit non-monotonic stress-strain behavior, modeled with an expansion to third order in volume strain, where the coefficients of the expansion depend on the metamaterial structure. The resulting constitutive behavior of interest displays regimes of both positive and negative stiffness and the inclusion therefore exhibits hysteretic snapping when forced by an acoustic pressure. Two cases are explored using a generalized Rayleigh-Plesset analysis to model the large-deformation dynamics of the metamaterial inclusion following an approach similar to Emelianov *et al.* [J. Acoust. Soc. Am., **115**, 581 (2004)]. The first case focuses on the forced dynamics of a single inclusion embedded in a weakly compressible elastic medium. The second case broadens the model to analyze the behavior of a heterogeneous material comprised of a low volume fraction of non-interacting metamaterial inclusions embedded in a weakly compressible material. Finally, estimates of the effective bulk modulus and loss factor of the heterogeneous medium are presented for instances of the forcing pressure inducing either large or small inclusion deformation. [Work supported by the ARL:UT McKinney Fellowship in Acoustics and the Office of Naval Research.]

2:35

2pSA6. Large-amplitude stress waves in nonlinear periodic structures. Pai Wang, Filippo Casadei, and Katia Bertoldi (SEAS, Harvard Univ., 29 Oxford St., Cambridge, MA, pai@seas.harvard.edu)

The ultimate goal of this research is to investigate the propagation of large-amplitude stress waves in nonlinear periodic structures. Sources of nonlinearity are associated with large-strain kinematics, material non-linearity, and bifurcation paths. In this study, we use a numerical approach to investigate the propagation on large strain waves in periodic lattice structures of finite size. Insights on the dispersion properties of such systems, and their functional dependence on the strain levels, are obtained by post-processing the time-history results obtained through time-domain explicit simulations. In particular, we highlight the effects of nonlinear amplitude parameters on the bandgaps and wave directionality of the considered systems.

2:50

2pSA7. Control of the dynamic properties of nano cantilevers with structural imperfections. Marcus Rutner, Dimitri Donskoy, and Mark Conticchio (Civil, Environ., and Ocean Eng., Stevens Inst. of Technol., Castle Point on Hudson, Hoboken, NJ 07030, mrutner@stevens.edu)

Acoustic metamaterials can be made out of micro/nano size structures employing various structural elements such as cantilever oscillators [JASA, **132**(4), 2866–2872]. Natural frequencies and eigenmodes depend on stiffness and mass distribution which should be reflected in macro as well as micro (nano) structure analysis. This study comprises three parallel approaches to model nano cantilever beams, i.e., the analytical method, the finite element analysis, and the molecular/atomic dynamics analysis, to identify how material imperfections influence the dynamic response of the nanocantilever. The study explores to what extent and under what circumstances macrostructure mechanics differs from nanostructure molecular/atomistic mechanics and how the built-in imperfection can be used to control structural dynamic properties at various scales.

Session 2pSC

Speech Communication: Determinants of Speech Perception: Session in Honor of Joanne L. Miller

Rachel M. Theodore, Cochair
Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269

Robert E. Remez, Cochair
Dept. of Psychol., Barnard College, Columbia Univ., 3009 Broadway, New York, NY 10027

Chair's Introduction—1:00

Invited Papers

1:05

2pSC1. Talker-specific influences on phonetic category structure. Rachel M. Theodore (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu)

A major goal of research in spoken language processing has been to describe how listeners achieve stable perception given the marked variability in mapping between the acoustic signal and linguistic representation. Toward this end, the research of Joanne L. Miller and colleagues has shown that speech sound categories, like other cognitive/perceptual categories, have rich internal structure, with category membership represented in a graded fashion. Moreover, category structure robustly shifts as a function of variation in the speech signal including variation associated with phonetic context, speaking rate, and dialect. I will discuss evidence indicating that internal structure also reflects talker-specific phonetic variation. These experiments concern talker differences in voice-onset-time (VOT), an acoustic parameter that marks the voicing distinction in stop consonants. I will begin with findings from production experiments that explicate talker differences in VOT with the goal of generating predictions for how listeners might accommodate such differences. I will then present findings from perception experiments indicating that listeners comprehensively adjust the mapping between the acoustic signal and phonetic category to reflect a talker's characteristic VOT distribution. Collectively, these findings demonstrate that listeners accommodate variability in the speech stream by dynamically adjusting internal category structure in light of systematic acoustic variation.

1:25

2pSC2. Measuring visual contributions in phonetic categorization. Lawrence Brancazio (Psych., Southern Connecticut State Univ., 501 Crescent St., New Haven, CT 06410, brancaziol1@southernct.edu)

Although most of Joanne Miller's work has explored the mapping from the acoustic signal to phonetic categories, she has also investigated the contributions of visual phonetic information in this process (Green and Miller, 1985). Audiovisual integration is commonly assessed using the McGurk effect, an illusion occurring with audiovisually conflicting stimuli. We (Brancazio and Miller, 2005) suggested that the McGurk effect may underestimate visual contributions in speech perception, in part based on our finding that visual speaking rate influences phonetic judgments when the McGurk effect does not occur (with stimuli that typically produce the effect). Rather, audiovisual integration of incongruent stimuli might result in percepts that fall between phonetic categories—including the one consistent with the acoustic signal—and then are mapped onto one of these categories. Thus, variability in the incidence of the McGurk effect might reflect variability in the process of mapping onto phonetic categories more so than variability in audiovisual integration. In recent work, I have sought to disentangle some factors that might influence the magnitude of the McGurk effect. I will describe findings from recent studies involving variations on the standard McGurk paradigm, and discuss the implications for developing finer-grained methods of assessing audiovisual integration.

1:45

2pSC3. Neural sensitivity to phonetic category structure. Emily Myers (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 850 Bolton Rd., Storrs Mansfield, CT 06269, emilybmyers@gmail.com)

Much of the recent work on the neural bases of speech perception has concentrated on the phenomenon of categorical perception, and in particular the observation that listeners appear to ignore or be insensitive to acoustic variation within the phonetic category. This has led to a search for phoneme-level processes in the brain, while neglecting a crucial aspect of the speech category, namely, that phonetic categories contain rich internal structure. Pioneering research on phonetic category structure from Joanne L. Miller and her colleagues shows us that listeners are not only sensitive to variation within the phonetic category, but that this sensitivity is modulated by speech rate, context, and talker identity. Recent work from our group using fMRI shows that the neural systems underlying phonetic processing are likewise sensitive to phonetic category structure, with neural responses in the temporal lobes that reflect to the "goodness

of fit" of a token to its phonetic category. In this paper I discuss evidence for the neural encoding of phonetic category structure, and in particular the sensitivity of this encoding to context and to experience.

2:05

2pSC4. Some neuromyths concerning the effectiveness of cochlear implants: Inferring function from dysfunction. David B. Pisoni (Psychol. and Brain Sci., Indiana Univ., 1101 E 10th St., Bloomington, IN 47405, pisoni@indiana.edu) and David B. Pisoni (Dept. of Otolaryngol. - HNS, Indiana Univ. School of Medicine, Bloomington, IN)

Cochlear implants provide profoundly deaf infants and young children with access to critical temporal and spectral patterns of speech needed for language development. Despite the enormous benefits of cochlear implants, many controversial questions remain. I will consider ten pressing issues that are the focus of current efforts in the field: (1) individual variability in speech and language outcomes, (2) neuroplasticity and early implantation, (3) learning and linguistic experience, (4) linguistic vs indexical channels in speech perception, (5) bilateral vs. unilateral implantation, (6) Contribution of other neural and cognitive systems, (7) early predictors of outcomes and risk factors, (8) workload and mental effort, (9) speech in noise, and (10) acoustic simulations in normal-hearing listeners. These issues raise important theoretical questions about basic processes in speech perception and spoken language processing. Cochlear implant research may be viewed as a "model system" enabling inferences about normal function from the study of dysfunction providing additional converging support for the proposal that the ear does not function in isolation from the rest of the brain; it is an inseparable component of a complex adaptive self-organizing system that evolved to support the perception and production of spoken language. [Work supported by NIH Grants: R01 DC-XYZ and DC-ABC to Indiana University.]

2:25

2pSC5. Effects of early auditory deprivation on auditory-visual development. Derek Houston (Otolaryngol. - Head & Neck Surgery, Indiana Univ. School of Medicine, 699 Riley Hospital Dr., RR044, Indianapolis, IN 46202, dmhousto@indiana.edu)

Auditory perception does not develop in isolation. In typically developing infants, the auditory system develops integrally with other sensory and motor systems. This integrality is disrupted in deaf infants. Even those who gain access to sound through cochlear implantation undergo a period of auditory deprivation where the other sensory systems develop independently from audition before access to sound begins. In this presentation, I will report findings from several studies of infants and toddlers with cochlear implants where we have found that they perform more poorly on auditory-visual integration and association tasks (e.g., novel word learning) than normal-hearing peers. Moreover, longer periods of auditory deprivation correlate with poorer performance on auditory-visual association tasks. I will discuss the implications of these findings on sensitive period of language acquisition. I will also present preliminary results from a new study that is further exploring multi-modal integration and learning in hearing-impaired infants. We are using head-mounted cameras, eye trackers, and microphones to analyze multi-modal communicative interactions between infants with hearing loss and their parents during free play sessions. By investigating the dynamics between multi-modal input and interactive behavior, we hope to gain important insights into how impaired sensory integration affects communicative interactions and multi-modal learning.

2:45

2pSC6. Understanding the fine structure of speech: Contributions of Joanne L. Miller. Patricia K. Kuhl (Inst. for Learning & Brain Sci., Univ. of Washington, Box 357920, Seattle, WA 98195, pkkuhl@u.washington.edu)

As a graduate student in the 1960s, Joanne Miller was extremely well organized, exceptionally attentive to details, and very goal directed. When Joanne directed that level of attention toward the fine structure of speech and to theories explaining how human perceivers (both adult and infant) deciphered it, many important discoveries were made. In this talk, I'll review some of the data and theoretical arguments that Joanne Miller has put forward, and show how they advanced the field.

3:05–3:30 Break

Contributed Papers

3:30

2pSC7. Imitation of a talker improves perception of the talker's speech. James W. Dias, Theresa C. Cook, Dominique C. Simmons, Josh J. Dorsi, and Lawrence D. Rosenblum (Psych., Univ. of California, Riverside, 900 University Ave., Riverside, CA 92521, jdias001@ucr.edu)

Human perceivers have a tendency to imitate the idiolect (talker-specific articulatory style) of a perceived talker. This phonetic convergence manifests between perceivers during live conversations and when perceivers shadow (say aloud) the speech spoken by a pre-recorded talker. The following investigation explores the potential facilitative effects of phonetic convergence on subsequent perception of the imitated talker's speech. To test this, the strength of phonetic convergence was manipulated by varying the delay of perceivers' shadowing responses. When perceivers are required to delay their shadowing responses, phonetic convergence has been found to reduce [Goldinger, Psychol. Rev. **105**(2), 215–279 (1998)]. If phonetic convergence can facilitate later perception of speech spoken by the shadowed talker, then perceivers who immediately shadow a talker should better identify the speech spoken by that talker, compared to perceivers who delayed their shadowing responses. Results suggest immediate—but not delayed—shadowing facilitates later identification of shadowed words spoken by the shadowed talker, compared to a non-shadowed talker. However, the facilitation effect does not transfer to utterances of novel (non-shadowed) words. The results may suggest a link between the mechanisms of speech perception and speech production based on idiolectic information available within lexical episodes.

3:45

2pSC8. Facilitating perception of speech in babble through conceptual relationships. Sara Guediche, Megan Reilly, and Sheila E. Blumstein (Dept. of Cognit., Linguistic, and Psychol. Sci., Brown Univ., 190 Thayer St., Providence, RI 02906, Sara_Guediche@brown.edu)

Speech perception is influenced by many different sources of information. Here we examine whether auditorily presented conceptual/semantic information facilitates the perception of degraded speech. To this end, acoustically clear sentences preceded sentences presented in speech babble. These sentence pairs were either conceptually related, conceptually

unrelated, or the same. The conceptually related and unrelated sentence pairs did not share any content words. Behavioral results show the highest accuracy for the same condition, followed by the conceptually related condition. The worst performance was in the unrelated condition. We then examined the neural substrates of this effect using fMRI. Preliminary results show that a direct contrast between related and unrelated sentences recruits a semantic/conceptual network including fronto-parietal and subcortical areas. The same sentence pair condition showed relatively greater activation in the superior temporal gyrus compared to the other two conditions. These findings suggest that the type of congruency (acoustic-phonetic vs. conceptual) enhances the activation of different functional networks. They also suggest that conceptually related sentences override a reliance on temporal lobe structures typically activated in resolving the perception of degraded speech.

Invited Papers

4:00

2pSC9. Dances of the tongue: Temporal cues and temporal context in the production and perception of vowels. Winifred Strange (Retired, P.O. Box 1226, Anna Maria, FL 34216, strangepin@aol.com)

One focus of the research program of Joanne Miller and her colleagues has been on the influence of temporal context (including speaking rate) on the perception of consonant contrasts differentiated primarily by temporal cues (e.g., Voice Onset Time as a major cue for voicing contrasts; formant transition durations for manner contrasts). This talk will summarize some of my own findings on the role of temporal context (speaking style and rate) in the perception of vowels by native and non-native speakers of American English (AE). In AE Consonant-Vowel-Consonant syllables, duration varies systematically, but redundantly, among vowels contrasting in vowel "height," cued primarily by first formant (F1) target frequency. Additional patterns of F1 temporal trajectories (timing of F1 maximum frequency and temporal symmetry/asymmetry of F1 onset and offset trajectories) differentiate so-called tense (long) and lax (short) AE vowels. Results of this research support the conception of speech production as a rhythmic activity with nested levels of timing control. Temporal patterns in syllabic- and multisyllabic-length segments of speech simultaneously provide information for phonetic identity and super-segmental structure within an exquisitely choreographed "dance" of the articulators. Differences in perception by native and non-native listeners support the conclusion that these are language-specific, learned patterns.

4:20

2pSC10. The role of systematic variation in speech perception. Lynne C. Nygaard (Dept. of Psych., Emory Univ., Atlanta, GA 30322, lnygaar@emory.edu)

A signature problem in our understanding of spoken language processing is the highly variable nature of the speech signal. The realization of linguistic form changes profoundly from utterance to utterance, individual speaker to individual speaker, and group of speakers to group of speakers. On the one hand, this high degree of variation is a problem for accounts of speech perception and spoken language comprehension. Listeners must maintain robust perceptual constancy in the face of the enormous variability in the instantiation of linguistic form. On the other hand, listeners are sensitive to the fine-grained structure of linguistic segments that signals differences among talkers and speaking styles. Variation is informative providing important cues to attributes of the individual speaker and social context. Empirical and theoretical work will be presented that attempts to reconcile the stability of speech perception with the informative nature of systematic variation. This work suggests that listeners both dynamically adapt to systematic changes in linguistic category structure and encode linguistically relevant variation in representations of spoken language. The findings suggest considerable behavioral and representational plasticity in speech perception and spoken language processing and highlight the importance of lawful variation for spoken communication.

4:40

2pSC11. Talker contingency in spoken communication. Robert E. Remez (Dept. of Psych., Program in Neurosci. & Behavior, Barnard College, Columbia Univ., 3009 Broadway, New York, NY 10027, remez@columbia.edu)

Those who speak the same language share its words, yet each spoken expression is unique, nonetheless. Research on the shared linguistic attributes perceived from highly varied physical acoustics has classically invoked the perturbations that drive expression from canonical phonemic form: coarticulation, articulatory rate variation, and anatomical scale variation. More recently, studies confirm that affect, dialect, idiolect, and idiosyncrasy also shape phonetic expression. Is the perception of this acoustic-phonetic variation prothetic, in which graded variation in physical acoustics creates similarly graded phonetic impressions? Some investigations of American English voicing encourage this view, though descriptive and perceptual studies also show that subphonemic phonetic expression can vary discontinuously in production and in perception. Overall, such studies of talker contingency in production and perception constrain claims of grammatical governance of articulation, and define new challenges of perceptual explanation.

5:00–5:30 Panel Discussion

Session 2pSP

Signal Processing in Acoustics: Session in Honor of William M. Carey II

Edmund Sullivan, Chair
Prometheus Inc., 21 Arnold Ave., Newport, RI 02840

Contributed Papers

1:30

2pSP1. Spatial coherence and radiated power for sound propagating in complex ocean environments. Timothy Duda (Woods Hole Oceanogr.Inst., AOPE Dept. MS 11,Woods Hole, MA 02543, tduda@whoi.edu)

Underwater sound propagation in areas of complex bathymetry, variable water masses (fronts), or variable stratification can result in detailed patterns of signal phase and amplitude. Here, we examine a few of the countless scenarios with numerical simulation. Synthetic horizontal arrays can be laid down within the acoustic fields produced with time-varying three-dimensional acoustic simulations. The power received at the phones of the arrays and the spatial structures of amplitude and phase can be used to create an estimate of array-exploitable power transmitted from a source at a known location to positions throughout the environment. Maps of this parameter and its components (horizontal coherence length and incoherent power arriving at the arrays) are presented for multiple simulated environments, including offshore of Southern California and offshore of the eastern seaboard of the United States.

1:45

2pSP2. Arrays and signal processing during the "Nantucket Sound Experiment": A review of work in honor of William M. Carey. Jason D. Holmes (Raytheon BBN Technologies, 10 Moulton St., Cambridge, MA 02375, jholmes@bbn.com)

Bill Carey was an engineer at heart and he loved finding practical solutions to important sonar problems. The development of a small hydrophone array towed by an unmanned vehicle built off of practical experience in one area of Bill's expertise (towed array signal processing and coherence length) and served as a practical tool to investigate another area he was interested in (acoustics of bottom interaction). This paper discusses what Bill referred to as the "Nantucket Sound Experiment" in which the small towed array was deployed. The development of the array, its use to measure sediment properties by forming a long synthetic aperture, and the relationship between synthetic array performance and coherence are discussed. Particular attention is paid to Bill's insight into the mechanisms that influenced array performance for various aspects of the experiment.

Invited Papers

2:00

2pSP3. The "Carey Number" continued—Mechanisms governing the horizontal array coherence length in shallow water acoustics. James F. Lynch, Timothy F. Duda, Ying-Tsong Lin, and Arthur Newhall (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 209 Bigelow Lab, MS #11, Woods Hole, MA, jlynch@whoi.edu)

The use of simplified "feature models" (geometric idealizations of specific, isolated ocean features) for coastal oceanographic features can allow one to calculate acoustically useful quantities approximately and even generate analytic forms for them. Feature models for coastal fronts, eddies, internal tides, linear and nonlinear internal waves, and spice are presented, and the scattering of sound from these objects is calculated. This allows one to estimate the useful quantity Lcoh, the horizontal coherence length that represents a physical limit for array signal processing. Continuing work on calculations of Lcoh, including source/receiver motion, will be presented. [Work sponsored by the Office of Naval Research.]

2:15

2pSP4. Bill Carey as a scientist, motivator, sponsor, and colleague. Thomas G. Muir, D. P. Knobles (Appl. Res. Labs., Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78713, muir@arlut.utexas.edu), and Clark Penrod (Appl. Res. Labs., Univ. of Texas at Austin, Austin, Vermont)

Bill Carey's long association with the Applied Research Laboratories at the University of Texas at Austin (ARL:UT), gave us the pleasure and good fortune to be associated with him in a wide range of pursuits, over many years. His insight and fortitude as a scientist and motivator led a number of us into productive scientific research that we may not have otherwise undertaken. As a sponsor at the Defense Advanced Research Projects Agency (DARPA), he posed visceral questions with unique viewpoints, sometimes in forceful terms, which led us to undertake difficult and productive projects. Some of these projects are linked to the series of shallow water sea trials called the Area Characterization Tests, which proposed and tested hypotheses, while collecting invaluable data to delineate the acoustics of significant scenarios in littoral ocean environments. We were privileged to participate in these experiments as well as analyze and model the results. As a colleague, while at other institutions, Bill provided the motivation and encouragement for us to undertake many projects, including a seismo-acoustic experiment with the SACLANT Undersea Research Center, which demonstrated the dispersive beam-forming concept for geophone arrays on the sea floor and for the detection and study of Sholte type interface waves. We pay tribute to Bill Carey in this talk by illustrating some of the discoveries made in a number of at-sea experiments and the theoretical and modeling work they spawned. [Work supported by ARL:UT Austin.]

2:30–2:45 Break

2:45

2pSP5. William M. Carey and the nonlinear frequency dependence of low frequency attenuation in sandy sediments. Allan D. Pierce (Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY), Jon M. Collis (Appl. Mathematics and Statistics, Colorado School of Mines, Golden, CO), Richard B. Evans (Retired, North Stonington, CT), and Jason Holmes (Sensing and Control Systems, Raytheon BBN Technologies, Cambridge, MA)

The early "conventional wisdom" was that low frequency sediment attenuation could be predicted with a downward extrapolation with attenuation presumed directly proportional to frequency. This goes back to work at Woods Hole reported in a 1968 Bryn Mawr doctoral thesis by Bennett. However, the attenuation at such low frequencies is difficult to measure directly. In subsequent years, a number of investigators, including Carey and colleagues, using a variety of experimental techniques (all indirect) different from that of Bennett, found frequency dependences with exponents substantially larger than unity. A survey paper by Holmes, Carey, Dediu, and Siegmann in 2007 concluded that the appropriate exponent was approximately 1.8. Carey, however, felt that a substantial part of the community still clung to the belief that the dependence was linear, and led an effort to garner support from basic theory. Analytical models, including that of Biot and others based on a rigorous examination of water-sand-pebble interaction at low frequencies, predicted that the attenuation should be quadratic in frequency. Carey conjectured that the discrepancy was because the inference technique ignored the possibility of shear waves in the sediment. That this explained the discrepancy was subsequently confirmed with computations by Collis and theoretical analysis by Pierce.

Contributed Paper

3:00

2pSP6. Comparison of shallow water mode transport theory with acoustic transmissions during Shallow Water Experiment 2006. Kaustubha Raghukumar, John A. Colosi (Oceanogr., Naval Postgrad. School, 315B Spanagel Hall, Monterey, CA 93943, kraghuku@nps.edu), Ying-Tsong Lin, Timothy F. Duda, Arthur Newhall (Appl. Ocean Phys. & Eng. Dept., Woods Hole Oceanogr. Inst., Woods Hole, MA), Kyle M. Becker (Appl. Res. Lab., Penn State Univ., State College, PA), and Paul Hines (Defence R&D Canada – Atlantic, Dartmouth, NS, Canada)

Coupled-mode transport theory appears to have put on solid theoretical ground acoustical scattering by internal waves in both deep and shallow water, over a range of low, medium, and high frequencies [Raghukumar and Colosi (2014) and Colosi and Morozov (2009)]. Here, full-field theoretical calculations of the acoustic field moments are compared against experimental data gathered during the Shallow Water Experiment 2006. Transport theory at low frequency is validated using data gathered by the WHOI Shark array at 175 Hz with a source towed by R/V Sharp at several different speeds over distances of 1.5–5.5 km. At high frequencies, comparisons are made at 1.2 kHz using data received by a WHOI bottom-mounted single hydrophone unit with a source towed by CFAV Quest over distances of 0–20 km. Acoustic observables include the mean and variance of intensity. The effect of a range dependent stochastic internal wave field is examined in the context of data-model comparison, along with the effect of random surface waves. In addition, further insights into mode coupling are presented using the shallow water hybrid transport theory.

Invited Paper

3:15

2pSP7. Bill Carey and passive synthetic aperture. Edmund Sullivan (Prometheus Inc., 21 Arnold Ave., Newport, RI 02840, ejsul@fastmail.fm)

Although it was long a controversial subject in the acoustics community, passive synthetic aperture remained of great interest to Bill Carey. Over the years, there were several "proofs" that it couldn't be done in any practical way. These proofs were technically correct in terms of the models upon which they were based, but it eventually became clear that these models were quite constraining in that they had no relation to actual practice and further, were hamstrung in that they were centered on the concept of beamforming. It is shown how passive synthetic aperture was placed on a firm theoretical basis by avoiding the focus on a synthetic "beam pattern" and treating the bearing and range estimation as pure estimation problems, where the approach is based on a joint estimation of bearing and source frequency, or in the case of the wavefront curvature problem a joint estimation of bearing, source frequency and range. A history of Bill's contributions to the area is outlined and explanations of the shortcomings of the so-called "proofs" are discussed. Several examples of experimentally verified results are outlined and several examples are given.

Meeting of Accredited Standards Committee (ASC) S3 Bioacoustics

C.J. Struck, Chair ASC S3
CJS Labs, 57 States Street, San Francisco, CA 94114-1401

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, *6* May 2014,

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.

TUESDAY AFTERNOON, 6 MAY 2014 PROVIDENCE 1, 3:45 P.M. TO 5:00 P.M.

Meeting of Accredited Standards Committee (ASC) S3/SC 1, Animal Bioacoustics

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D.K. Delaney, Chair ASC S3/SC 1
USA CERL, 2902 Newmark Drive, Champaign, IL 61822

D.S. Houser, Vice Chair ASC S3/SC 1
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, 6 May 2014.

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 EVENING, 6 MAY 2014 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, Wednesday, and Thursday evenings.

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:

4:30 p.m.	Engineering Acoustics	550AB
7:30 p.m.	Acoustical Oceanography	552AB
7:30 p.m.	Architectural Acoustics	555AAB
7:30 p.m.	Physical Acoustics	551AB
7:30 p.m.	Psychological and Physiological Acoustics	554AB
8:00 p.m.	Structural Acoustics and Vibration	553AB

Session 3aAAa

Architectural Acoustics: J. Christopher Jaffe—His Life in Acoustics

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

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

Alexander U. Case, Cochair
Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854

Chair's Introduction—8:25

Invited Papers

8:30

3aAAa1. Recollections on Chris Jaffe: From his early days in ASA through an enduring friendship of nearly six decades. William J. Cavanaugh (Cavanaugh Tocci Assoc. Inc., 327F Boston Post Rd., Sudbury, MA 01776, wcavanaugh@cavtocci.com)

In some 60 years of consulting in architectural and environmental acoustics, one encounters few colleagues who fit the definition of a truly renaissance man. J. Christopher Jaffe is one of those few. His love for the performing arts and in designing facilities for the enjoyment of countless audiences and performers alike knew no bounds. My friendship with Chris began with hearing the very first paper he presented at an ASA meeting on the frequency selective properties of stage enclosures and extended throughout his entire professional career. Chris Jaffe epitomizes the challenge stated in the by-laws of our Society: to spread knowledge in acoustics and to promote its practical applications. Chris lived this challenge for a too short professional life with unabated enthusiasm and joy. We miss him terribly but are comforted that he made the world a better place.

8:45

3aAAa2. J. Christopher Jaffe: Scientist and friend. Leo L. Beranek (Retired, 10 Longwood Dr., Westwood, MA 02090, beranekleo@ieee.org)

Chris Jaffe is acclaimed for his original designs of spaces for the performance of music. It has been this author/s pleasure to discuss his many projects and to hear concerts in his four best known halls: Severance Hall in Cleveland, Ohio; Bass Performance Hall in Fort Worth, Texas; the Concert Hall at the Kennedy Center in Washington, D.C.; and the Sala Nezahualcoyotln in Mexico City. Chris took great pleasure in teaching others the field and was the leader in establishing the Concert Hall Research Group that took data in many halls and held seminars for acousticians at all levels. Other facets in his life will be discussed.

9:00

3aAAa3. Remembrances of Chris Jaffe—An innovator in many fields, our colleague, and friend. Wade Bray (HEAD Acoust., Inc., 6964 Kensington Rd., Brighton, MI 48116, wbray@headacoustics.com), Mahlon D. Burkhard (Retired, Adamstown, MD), and Klaus Genuit (HEAD Acoust. GmbH, Herzogenrath, Nordrhein-Westfalia, Germany)

A remembrance of Christopher Jaffe in the technical perspective will be given, in a matrix of the human perspective of working with him, learning from him and enjoying his vision, friendship, and wit. The authors all collaborated with Chris: Wade Bray as a consultant at Jaffe Acoustics, Mahlon Burkhard as an acoustical colleague and developer of electronic technologies proposed by Chris through the major era of the Electronic Reflected Energy System (ERES) starting with the "Live from Studio 8H" NBC television symphonic concerts—birth of the Chris-named "NBC Delay"—and Klaus Genuit fulfilling Chris's vision of binaural technology in multiple uses—Chris providing the avenue for HEAD acoustics GmbH to begin serving North America. Christopher Jaffe was a thorough, novel acoustician: fluent and innovative in both non-electronic and electronic techniques, though perhaps better known for his lifelong passion about the latter in the pure service of the listening experience and its reaching more people. He had a unique personal, charismatic presence, an ability not only to explain technical acoustics in artistic context to nontechnical people, but also to enfold them in the enthusiasm, skill and momentum of his spirit. We celebrate him; we miss him, his path lives.

9:15

3aAAa4. Chris Jaffe's contributions to the Concert Hall Research Group. Timothy Foulkes (Cavanaugh Tocci Assoc., 327 Boston Post Rd., Sudbury, MA 01776, tfoulkes@cavtocci.com)

One of Chris Jaffe's many contributions to acoustics was his involvement in the Concert Hall Research Group (CHRG). In 1992, Chris had the idea of sending three measurement teams to the same concert halls for the dual purpose of getting current data and also to see how close the results would be between different teams measuring the same hall. From the inception of the CHRG in 1992 until a few months before his death in 2012, Chris was a source of ideas, inspiration, and energy.

9:30

3aAAa5. The Jaffe effect. Robin S. Glosemeyer Petrone (Threshold Acoust., 53 W Jackson Blvd., Ste. 815, Chicago, IL 60604, robin@thresholdacoustics.com)

By 1940, nearly every sizable city in America had a movie palace or movie-adapted vaudeville theater. As these regal houses for the silver screen were replaced by cinemas, the palaces and vaudeville theaters of the early 20th century sat with empty stages, taunting orchestras, theater companies, and would be audiences. Christopher Jaffe's most noted contribution to architectural acoustics was championing the adaptive re-use of these theaters through the development of an orchestra shell system. These light-weight shell systems allowed the theaters to support multiple performing arts organizations without a monumental capital campaign that neither the cities nor art organizations could afford. Chris' most singular contribution to acoustics was, however, his mentorship. This maverick in the field, always on a quest to prove himself, never missed an opportunity to share his knowledge or work out a new concept with a colleague or apprentice. He fostered the growth of consultants at all stages of their careers, first in his practice and later in the establishment of an architectural acoustics program at his alma mater.

9:45

3aAAa6. Christopher Jaffe and the Graduate Program in Architectural Acoustics at Rensselaer Polytechnic Institute. Ning Xiang (School of Architecture, Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, xiangn@rpi.edu)

Chris Jaffe graduated with a major in chemical engineering in 1949 from Rensselaer Polytechnic Institute. Four decades later in 1998, Chris founded an Architectural Acoustics program at the School of Architecture, Rensselaer Polytechnic Institute. In the fields of architectural-, physical-, and psycho-acoustics, the rapid pace of change has advanced the program to be a graduate program with an ambitious mission of educating future experts and leaders in architectural acoustics. Chris Jaffe's continued dedication and support helped Rensselaer's Graduate Program in Architectural Acoustics reshape its pedagogy using "STEM" (science, technology, engineering, and mathematics) methods, including intensive, integrative hands-on experimental components that fuse theory and practice in a collaborative environment. The STEM-based pedagogy enables individuals from a broad range of fields to succeed in this rapidly changing field. The program has attracted graduate students from a variety of disciplines including individuals with B.Arch., B.S., or B.A. degrees in Architecture, Music, Engineering, Audio/Recording Engineering, Physics, Mathematics, Computer Science, Acoustics, Electronic Media, and related fields. RPI's Graduate Program in Architectural Acoustics has since graduated more than 100 graduates with both M.S. and Ph.D. degrees. This paper shares the growth and evolution of the graduate program, and acknowledges the profound contributions made by Chris Jaffe.

10:00–10:15 Break

10:15

3aAAa7. Jaffe's signature: Science, with art. Alexander U. Case (Sound Recording Technol., Univ. of Massachusetts Lowell, 35 Wilder St., Ste. 3, Lowell, MA 01854, alex@fermata.biz)

I am one of a handful of students claiming the title, "Chris Jaffe's first graduate student,"—but I really am the first! I found at Rensselaer Polytechnic Institute a program that reflects his priorities. We did not just study the physics of room acoustics and make measurements in built spaces. We worked side by side with architects, engineers, electronic musicians, and other artists. Acoustics was not an isolated achievement. It was always part of a collaborative effort with other ambitious disciplines. We built things—big, awe-inspiring things—visually stunning, sonically thrilling. I found myself studying, measuring, and recording in all kinds of spaces, from concert halls in Texas to subway stations in New York. Thanks to Dr. Jaffe's leadership, were always doing, growing, learning, and laughing.

10:30

3aAAa8. Fortuitous coupling: A recollection of how Chris Jaffe brought together coupled rooms and Texas oil money to launch a graduate program and a career. Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

Early in his career Chris Jaffe discovered that reverberation-starved Vaudeville theaters could be improved through use of light-weight demountable shells for the stage enclosures. These shells could be designed for frequency-dependent "selective transmission" of energy into the otherwise unused stage house, realizing a "fortuitous coupling" that improved acoustics for the audience and on-stage musicians. My experience with Chris at Rensselaer Polytechnic Institute (RPI) is also one of fortuitous coupling, as he orchestrated the launch a new graduate program in architectural acoustics. Chris had just overseen the completion of Bass Performance Hall in Fort Worth, Texas, which realized the culmination of his design philosophy for stage-house coupling in multipurpose theaters. Now, Bass Hall was to become a laboratory for studying the science underlying Chris's empirical findings. RPI, Chris's alma mater, was to be the institutional home for that research. And Ed Bass, scion of the Bass family and primary benefactor of the hall's construction, would provide the initial support. As a student, about to complete my Master's degree and ready leave physics for audio engineering, a chance elective course followed by a Medici-esque offer to study the physics of coupled rooms radically shifted the course of my professional life.

10:45

3aAAa9. Chris Jaffe: Youthful golden years. Benjamin Markham and Carl Rosenberg (Acentech Inc., 33 Moulton St., Cambridge, MA 02138, bmarkham@acentech.com)

From 2009 until his demise—roughly 50 years after his entrée into the acoustics field—Chris Jaffe was a valued partner in the Acentech Studio A team. We will share the manner in which his mentorship and guidance informed us and our profession. His self-deprecating humor could not disguise his intense knowledge, broad experience, willingness to help mentor junior staff, and engagement in new endeavors. He imbued his acoustics consulting "golden years" with youthful vigor, and he accomplished much: he finished his book, consulted on projects, shared back stories, enlivened our parties, introduced us to his friends and his manner of collaborating with them, and made an indelible imprint on both the content and style of our performing arts practice. Our presentation will share these experiences as life lessons for success and respect in our profession.

11:00

3aAAa10. Chris Jaffe's three A's. Malcolm Holzman (Holzman Moss Bottino Architecture, 214 West 29th St. Tower 17fl, New York, NY 10001, mholzman@holzmanmoss.com)

The first A is for Academia: The importance Chris Jaffe put on education as a foundation and a concentration for his career. The second A is Acoustics: A lifespan focused on propelling the science and art of architectural acoustics in the civic and academic presentations of music, theater, and dance. The third A is Adirondacks: His affection and attachment to the Adirondacks were a measure of the man.

11:15–11:45 Open mic Discussion

WEDNESDAY MORNING, 7 MAY 2014 550 A/B, 9:00 A.M. TO 12:00 NOON

Session 3aAAb

Listening to the "Virtual Paul's Cross"—Auralizing 17th Century London I

Matthew Azevedo, Chair
Acentech Inc., 33 Moulton St., Cambridge, MA 02138

The purpose of this session is to provide an opportunity for people to listen to the Virtual Paul's Cross auralization, which allows listeners to experience John Donne's 1622 Gunpowder Day sermon while surrounded in three dimensions by a reactive crowd of up to five thousand, the bells of St. Paul's, and the ambient soundscape of 17th century London. The auralization allows for real-time changes to crowd size, listener position, the behavior of the intelligent agents which create the crowd reactions, and variations in the type and frequency of ambient sounds and requires over one hundred concurrent audio channels, a dozen channels of real-time convolution, hours of project-specific source recordings, and a complex network of intelligent and stochastic logical structures.

WEDNESDAY MORNING, 7 MAY 2014 554 A/B, 9:00 A.M. TO 12:00 NOON

Session 3aAB

Animal Bioacoustics: Sound Production and Reception by Animals

James A. Simmons, Chair
Neuroscience, Brown Univ., 185 Meeting St., Box GL-N, Providence, RI 02912

Contributed Papers

9:00

3aAB1. Behavioral analysis of lateral line and vestibular hair cell function in developing *Xenopus laevis*. Andrew T. Stevens-Smith and Andrea Simmons (Brown Univ., Providence, RI 02912, Andrew_T_Smith@brown.edu)

Xenopus tadpoles are aquatic, nocturnal animals with well-developed superficial lateral line neuromasts. We previously showed that these animals, from premetamorphic through early froglet stages, exhibit stereotyped responses to low velocity flow fields—they move downstream at flow onset, and then turn and orient towards the flow source (positive rheotaxis). They continue to station hold in this oriented position for the duration of the stimulus. Positive rheotaxis and station holding are disrupted, but not totally eliminated, when animals are exposed to high concentrations of the ototoxic drug gentamicin. These data suggest that another sensory system is involved in detection of current flow. To determine the role of vestibular hair cells in these behaviors, we injected gentamicin directly into the tadpoles' otic capsules. This procedure damaged the developing otoliths and resulted in abnormal circular swimming movements and increased latency of rheotaxis and station holding. These behaviors were not, however, completely disrupted. These data suggest that the lateral line and vestibular systems act together in mediating responses of Xenopus tadpoles to flow fields.

9:15

3aAB2. Selection on aerial hearing in turtles: Auditory evoked potentials in the box turtle, *Terrapene carolina* relative to the stinkpot, *Sternotherus odoratus*. Jeffrey Zeyl and Carol E. Johnston (Fisheries, Aquaculture and Aquatic Sci., Auburn Univ., 203 Swingle Hall, Auburn University, Auburn, AL 36849, jnz0002@tigermail.auburn.edu)

Mechanisms of auditory stimulation differ underwater versus in air due to differences in each medium's acoustic impedance, which has resulted in unique hearing adaptations in aquatic relative to terrestrial organisms. Testudines are a useful taxon for studying the evolution of hearing specializations in relation to the air-water interface because this group includes members at various points on the aquatic-terrestrial lifestyle continuum. Here we tested for differences in auditory function between the terrestrial box turtle, *Terrapene carolina* (Emydidae) and the fully aquatic stinkpot, *Sternotherus odoratus* (Kinosternidae). Auditory evoked potentials were collected in response to tone pips to generate threshold audiograms in air as well as with tympana submerged underwater. Sensitivities and bandwidths of both species were similar underwater and in air, but the thresholds of box turtles were 9–20 dB more sensitive than stinkpots in air across the entire frequency range. The results indicate selective pressures to enhance aerial hearing in box turtles.

9:30

3aAB3. Underwater signal attenuation of northern red-legged frog calls. Jodi Gronborg (Biology, Portland State Univ., 1719 SW 10th Ave., SRTC 246, Portland, OR 97201, gronborg@pdx.edu)

Shallow water acoustics can dramatically alter spectral profiles of the northern red-legged frog underwater advertisement calls and should be taken into consideration when designing man-made aquatic environments as part of habitat mitigation. While much is known about atmospheric constraints on acoustic communication, we need to learn more about aquatic constraints. What is known about underwater acoustic signal transmission is garnered primarily from deep ocean research, leaving much to be discovered about the relatively extreme shallow underwater environments such as exists in ponds. The northern red-legged frog, *Rana aurora*, is one of three ranid species shown to vocalize while submerged underwater, and only one of two known to use the underwater portion of its environment for mating. My hydrophone recordings of the underwater chorus include calls with dominant frequencies in the 5000—15000 Hz range, sharply contrasting with the established dominant frequency range for advertisement calls of the northern red-legged frog of 450–1300 Hz. I conducted frequency sweep playbacks at breeding which demonstrate that shallow residential water bodies have unique frequency responses that dramatically alter the spectral profiles of underwater signals. This changes the frequency characteristics of what is heard by females and could significantly impact reproductive success.

9:45

3aAB4. Stimulus-frequency and response timing in clouded leopards: Evidence for inner ear adaptation. Edward J. Walsh (Developmental Auditory Physiol. Lab, Boys Town National Res. Hospital, 555 North 30th St., Omaha, NE 68131, edward.walsh@boystown.org), Christopher A. Shera (Eaton-Peabody Labs., Harvard Med. School Massachusetts Eye & Ear Infirmary, Boston, MA), Carolina Abdala (Dept. of Otolaryngol., Univ. of Southern California, Los Angeles, CA), Heather E. Robertson (Nashville Zoo at Grassmere, Nashville, TN), and JoAnn McGee (Developmental Auditory Physiol. Lab, Boys Town National Res. Hospital, Omaha, NE)

Unlike the inverse relationship between response timing (latency) and stimulus frequency commonly observed in other fields, the relationship in species belonging to the Pantherinae subfamily is more complex. Although an inverse relationship exists between neural timing and low frequency stimuli (between 0.5 and 1–2 kHz), the relationship for frequencies greater than 2 kHz breaks from this pattern and is best described by a positive-going parabola-like curve with a maximum value in the vicinity of 8 to 16 kHz in representatives of the Pantherinae lineage studied thus far. In clouded leopards the local maximum is near 8 kHz. This general latency-frequency pattern has been confirmed in tigers and appears to hold for jaguars and lions. Clouded leopards (*Neofilis nebulosa*) branched directly from the Panthera lineage approximately 6 million years ago. This phylogenetic proximity to Panthera makes the genus Neofilis particularly interesting in relation to the response timing question. The outcome of previous efforts to determine the cochlear site of origin of short latency responses to lower frequency stimuli in the tiger were consistent with the existence of a basal turn timing adaptation. The implications of such adaptation in clouded leopards will be considered.

10:00

3aAB5. A subtraction technique for removing playback noise from high-frequency rodent recordings. Mustafa Z. Abbasi (Appl. Res. Lab. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78751, mustafa_abbasi@utexas.edu), Bret Pasch, Amilia Humber, Michael J. Ryan (Dept. of Integrative Biology, Univ. of Texas at Austin, Austin, TX), and Preston S. Wilson (Appl. Res. Lab. and Dept. of Mech. Eng., The Univ. of Texas at Austin, Austin, TX)

The efficacy of animal communication often necessitates signalers to adjust signals. For example, many vertebrates modify vocal output to minimize interference from ambient noise (e.g., Lombard effect) or in response to other signaling animals. Such auditory-feedback-mediated vocal control is well documented in songbirds, humans, and nonhuman primates, but has not been explored in rodents. Alston's singing mouse (*Scotinomys teguina*) emit advertisement vocalizations that function in mate attraction and male-male aggression, and can vary both temporal and spectral features depending on the social context. In this experiment, we investigated the extent to which mice can modify vocal output in response to perturbations in auditory feedback by broadcasting conspecific vocalizations that overlapped a focal male's song. However, an unexpected challenge was found in separating the mouse's vocalization from the broadcast stimuli. Other studies have used Golay codes to measure the impulse response of the system and subtract the noise. However, such techniques could not be applied herein due to the high frequencies produced by mice; movement of the mouses' head appears to change the system sufficiently to prevent effective subtraction. The authors will present a novel method using spectral envelopes and cross-correlation procedures to garner feedback on the validity of this technique.

10:15–10:30 Break

10:30

3aAB6. Frequency and intensity difference limens in mice. Katrina Toal, Kelly E. Radziwon, and Micheal L. Dent (Psych., Univ. at Buffalo, SUNY, 231 Affinity Ln., Apt. F, Buffalo, NY 14215, ktoal@buffalo.edu),

The ability to distinguish between different frequencies and intensities is a fundamental auditory process, yet it has not been explored in CBA/CaJ mice. Previous researchers have used behavioral approaches to examine frequency and intensity discrimination abilities of the NMRI mouse, feral house mice, and other strains of mice with known hearing impairments. The

present experiments use operant conditioning procedures to investigate the basic hearing abilities of the normal-hearing CBA/CaJ mouse. Intensity difference limens (IDLs) were obtained for six subjects and frequency difference limens (FDLs) were obtained for three subjects. For both experiments, the Method of Constant Stimuli and a threshold d' of 1.5 were used. IDLs and FDLs were obtained for 12, 16, 24, and 42 kHz tones. FDLs were obtained at 10 dB SL and 30 dB SL whereas the background for the IDL task was only 10 dB SL. At higher frequencies, the calculated FDLs increased. Furthermore, the thresholds were higher when sounds were presented at 30 dB SL compared to 10 dB SL. In the FDL experiment, the mice had a mean just-noticeable-difference of 3.5% Weber fraction across all four frequencies and sound levels. Interestingly, IDLs were similar across all frequencies.

10:45

3aAB7. Twists and turns, in cochlear anatomy: Curvatures related to infra vs ultrasonic hearing. Darlene R. Ketten (CMST, Appl. Phys. and Imaging, Curtin Univ./Harvard Med. School, CMST, GPO Box 1987, Perth, WA 6845, Australia, dketten@whoi.edu), James Simmons (NeuroSci., Brown Univ., Providence, RI), Hiroshi Riquimaroux (Graduate School of Life and Medical Sci., Doshisha Univ., Kyoto, Japan), Scott Cramer and Julie Arruda (Biology, Woods Hole Oceanogr. Inst. , Woods Hole, MA)

Microchiropteran bats and odontocete cetaceans are sophisticated echolocators with acute ultrasonic hearing operating in radically different media. Similarly, elephants and mysticetes share the ability to generate and respond to infrasonics. In this study, the heads, outer, middle, and inner ears of 32 specimens from 11 species of bats, dolphins, elephants, and whales were analyzed with microCT (11 to 100 micron isotropic voxel imaging; Siemens Volume Zoom and X-Tek CT units). Canal length, basilar membrane dimensions, and cochlear curvatures varied widely among all species. Length correlates with body mass, not hearing ranges. High and low frequency limits correlate with basilar membrane ratios and radii ratios, which are a measure of the radius of curvature. The ears of the known echolocators were significantly different from the mid to low frequency ears, with increased stiffness, thicker membranes, and outer osseous laminae supporting up to 60% of the basilar membrane. Anatomical correlates of "foveal" regions with stretched representation for peak echolocation spectra were found in both bat and porpoise ears. Radii and membrane ratios are consistent despite media and are predictive of high and low frequency hearing limits in all ear types. [Work supported by NIH, JIP, N45/LMRS -US Navy Environmental Division, and ONR Global.]

11:00

3aAB8. Enhance beam formation by airsacs and skull in Chinese river dolphin (*Lipotes vexillifer*). Chong Wei (College of Ocean & Earth Sci., Xiamen Univ., Hawaii Inst. of Marine Biology, Lilipuna Rd., Kaneohe, Hawaii 96744, chongwei@hawaii.com), Whitlow Au (Hawaii Inst. of Marine Biology, Univ. of Hawaii at Manoa, Kaneohe, HI), Zhongchang Song and Yu Zhang (Key Lab. of Underwater Acoust. Commun. and Marine Information Technol. of the Ministry of Education, Xiamen Univ., Xiamen, Fujian, China)

The melon of dolphins is considered by many as the structure responsible for the focusing of the biosonar beam. However, finite element numerical simulation of the head of the Chinese river dolphin (*Lipotes vexillifer*) indicates that the biosonar beam is formed by reflections off the airsacs and bony structures in the skull. The finite element approach was applied to numerically simulate the acoustic propagation through dolphin's head in four several situations (complete head, skull only, skull plus melon, and skull plus airsacs). The acoustic intensity distribution and the corresponding polar plots showed that the melon causes the beam to narrow slightly and affects the angle of the main beam. The airsacs kept the sound propagating to the anterior and focused the energy into the main lobe. The bony structure prevented the sound from propagating below the rostrum and contribute to energy in the main beam. The results suggest that the airsacs and the complex bony structure play a dominant role in the formation of the biosonar beam of a dolphin, more so than the melon.

11:15

3aAB9. *In vivo* ultrasonic attenuation in cetacean extracranial soft tissues. Michael D. Gray, Peter H. Rogers, Peter J. Cameron (School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332-0405, michael.gray@me.gatech.edu), and Gregory D. Bossart (Georgia Aquarium, Atlanta, GA)

In vivo ultrasonic attenuation was estimated for extracranial soft tissues of two *Tursiops truncatus* and one *Delphinapterus leucas*. Backscatter data were non-invasively collected as part of routine health-based ultrasound examinations using a transducer operating in the 2.0–3.5 MHz frequency range. Data sets collected over the proximal mandible and temporal regions were processed to yield estimates of attenuation using a reference tissue phantom whose properties had been independently determined. The estimated attenuations were at the low end of the range of reported values for *in vitro* mammalian fatty and connective tissues.

11:30

3aAB10. Numerical simulation of sound generation by Cicada. Derke Hughes (Sensors and SONAR, NUWCDIVNPT, 1176 Howell St., Newport, RI 02841, derke.hughes@navy.mil), Allan D. Pierce (Retired, East Sandwich, MA), Richard A. Katz (Sensor and SONAR, NUWCDIVNPT, Newport, RI), and Robert M. Koch (Chief Technol. Office, NUWC, Newport, RI)

The principal anatomical structures in the cicada that radiate sound are two platelets referred to as tymbals, which vibrate after being struck by ribs that have undergone buckling. This research effort investigates the sound of these ribbed finite plates connected to a parallel surface by a nonlinear spring. When individual ribs are placed under compression, the linearized version of the model predicts eventual exponential growth of the transverse displacement when the compressional load exceeds the buckling load. The nonlinear spring, however, stops this growth and a subsequent oscillation ensues. The actual anatomy of the cicada is more complicated than this basic model. However, this simplified mathematical explanation is given as a means to describe sound emitted in a sequence of closely spaced tone bursts. The energy from these sound impulses are stored in tensed muscles and released via buckling into the kinetic energy of ribs, which is similar to striking a drum. The tymbals "ring" at a frequency controlled by the mass of the tymbals and the air cavity "spring" within the abdomen. This ringing vibration affects the amplitude, cycles within each pulse, and the damping of the tymbal function to generate the efficiency of the cicada sound radiation.

11:45

3aAB11. A comparison of hees and haws: Donkey, Grevy's zebra, and African penguin. David Browning (139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com) and Peter Scheifele (Dept. of Comm. Sci., Univ. of Cincinnati, Cincinnati, OH)

A surprisingly few creatures bray (vocalizing during both breath in and breath out). It is not apparent why this is so or what advantage it gives those that do. This is a comparison of the technique and acoustic content for the three most well-known brayers. Donkeys (*Equus asinus*) have the most rich or raucus bray, depending on your point of view. Some donkeys start with a hee (breath inflow) and others with a haw (breath outflow), generally continuing until they are out of breath. The Grevy's zebra (*Equus grevy*) shares the perrisodactyl ability to vary frequency during vocalization but it appears to be a strained activity, resulting in a scaled down version of the donkey bray. The African penguin (*Spheniscus demersus*) has the most uniform and tonal haw, in most cases preceded by a series of short hees, apparently to increase breath.

Session 3aBAa

Biomedical Acoustics: Measurement and Imaging

Ronald A. Roy, Chair
Dept. of Eng. Sci., Univ. of Oxford, Oxford OX1 3PJ, UK

Contributed Papers

8:00

3aBAa1. Ultrasound acoustic shadow width is an accurate predictor of kidney stone size. Franklin C. Lee (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Barbrina Dunmire (Ctr. Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, Seattle, WA), Jonathan D. Harper (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Bryan W. Cunitz, Marla Paun, Michael Bailey (Ctr. Industrial and Medical Ultrasound, Appl. Phys. Lab, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, bailey@apl.washington.edu), and Mathew D. Sorensen (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA)

Previous studies have shown overestimation of kidney stone size in ultrasound images. We explored measuring the stone's acoustic shadow as a predictor of stone size. Forty-five calcium oxalate monohydrate (COM) kidney stones ranging from 1 to 10 mm were imaged in a water bath using a research-based ultrasound system and C5-2 transducer. Stones were imaged at depths of 6, 10, and 14 cm. The widths across the stone image and across acoustic shadow distal to the stone image were measured by the operator and through an automated algorithm. Measuring size across the stone image consistently overestimated: overestimation was 0.9±0.8 mm, 1.5±1.0 mm, 2.0±1.2 mm (manual) and 0.5±1.7 mm, 0.4±1.5 mm, 0.8±1.1 mm (automated) at 6, 10, and 14 cm depths. Measurement of the acoustic shadow width more accurately estimated stone size: 0.0±0.4 mm, 0.0±0.6 mm, and-0.2±0.8 mm (manual) and 0.2±0.5 mm, 0.1±0.8 mm, and 0.1±1.0 mm (automated) at 6, 10, and 14 cm depths. Measurement from the shadow reduced misclassification of passable stones <5 mm to requiring surgery >5 mm from 25% to 7%. The results have implications for directing treatment of asymptomatic stones based on ultrasound images. [Work supported by NIH DK043881, DK092197, and NSBRI through NASA NCC 9-58.]

8:15

3aBAa2. Biofilm mitigation by ultrasound-assisted liposome treatment applied to a national aeronautics and space administration project. Junru Wu (Phys., Univ. of Vermont, 1 Whiteface St., South Burlington, VT 05403, jwu@uvm.edu), Dong Ma (School of Biological Sci. and Medical Eng., Southeast Univ., Burlington, Vermont), Graham Willsey, and Matthew Wargo (Microbiology and Molecular Genetictons, Univ. of Vermont, Burlington, VT)

Space exploration requires an effective means of biofilm control for water reclamation systems that minimizes use of non-recyclable chemicals (such as iodine and antibiotics) and does not contaminate water in the water treatment system. Uncontrolled biofilm growth due to ineffective mitigation has been the cause of water reclamation system failures both on the Russian Mir Space Station and on the International Space Station, as well as corrosion problems in thermal systems. The challenge of biofilm migration mainly comes from: its matrix of extracellular polymeric substances, in which biofilm organisms embed themselves, inhabiting bacterium in biofilm treated by chemical and mechanical stresses. The goal of the project is to develop a new biofilm mitigation approach using targeted liposomes, which encapsulate an anti-microbial chemical agent, assisted by ultrasound to enhance mixing and attachment rate of the liposomes to the bacterial colonies within the biofilm. In this presentation, we will present some preliminary results of enhancement of penetration of liposomes of nanometer scales into biofilms using mild nonfocused ultrasound. (The spatial average-temporal average power and intensity are 15.1 W and 2.99 W/cm^2, respectively, duty cycle = 50%.)

8:30

3aBAa3. Treatment planning and strategies for acousto-optic guided high-intensity focused ultrasound therapies. Matthew T. Adams (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, adamsm2@bu.edu), Robin O. Cleveland (Inst. of Biomedical Eng., Univ. of Oxford, Oxford, United Kingdom), and Ronald A. Roy (Dept. of Eng. Sci., Univ. of Oxford, Oxford, United Kingdom)

Real-time acousto-optic (AO) sensing has been shown to non-invasively detect changes in *ex vivo* tissue optical properties during high-intensity focused ultrasound (HIFU) exposures. The technique is particularly appropriate for monitoring non-cavitating lesions that offer minimal acoustic contrast. This work employs a modeling-based approach to improve the AO sensing of lesion formation during HIFU therapy, to develop treatment strategies for the ablation of large volumes, and to assess the technique's viability and robustness in a clinical setting. The angular spectrum method is used to model the acoustic field from the HIFU source. Spatio-temporal temperature elevations induced by the absorption of ultrasound are modeled using a finite-difference time-domain solution to the Pennes bioheat equation. Changes in tissue optical properties are calculated using a thermal dose model, calibrated using experimental data. The diffuse optical field is modeled using an open-source GPU-accelerated Monte Carlo algorithm. The Monte Carlo algorithm is modified to account for light-sound interactions, using the acoustic field from the angular spectrum method, and to account for AO signal detection. AO signals are presented in the context of a photorefractive-crystal-based detection scheme, and are compared to signals obtained using standard optical detectors. [Work supported in part by the Whitaker International Program.]

8:45

3aBAa4. Using high-frequency ultrasound to detect cytoskeletal dysfunction in Alzheimer's disease. Ashley N. Calder, Janice E. Sugiyama (Biology, Utah Valley Univ., 800 W. University Parkway, Orem, UT 84058-5999, ashleycalder09@gmail.com), Laurel A. Thompson (Chemistry, Utah Valley Univ., Orem, UT), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Recent work indicates that Alzheimer's disease (AD) affects the cytoskeleton and cellular structure through mutations that alter structural proteins, and that dysfunction of the cytoskeleton may play a pivotal role in AD and other neurodegenerative diseases. The goal of our research is to determine if high-frequency ultrasound can detect cytoskeletal dysfunction in AD. Research on the molecular subtypes of breast cancer indicate that mutations specific to each subtype may change the characteristics of the cytoskeleton and resulting properties of the cell such as size, shape, and stiffness. Both computer simulation and experiment have demonstrated that high-frequency ultrasound in the 10–100 MHz range is sensitive to these properties. For this study, ultrasonic tests were conducted on monolayer cell cultures of breast cancer cell lines of different subtypes. The ultrasonic spectra were

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compared and correlated to model results using a pattern recognition algorithm. Preliminary results indicate that cell stiffness and size can be determined from the measurements. The cytoskeletal properties of the cells were additionally modified by chemical and physical agents such as the introduction of colchicine and electric fields to mimic the effects of AD. Results from these and future studies with neuron cell cultures will be discussed.

9:00

3aBAa5. Molecular subtyping of colorectal cancer using high-frequency ultrasound. Alexis M. Holman (Biology, Utah Valley Univ., 800 W. University Parkway, Orem, UT 84058-5999, mgn.alexis@gmail.com) and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

Previous studies have shown high-frequency (HF) ultrasound (10–100 MHz) may be sufficiently sensitive to detect and differentiate between both the histopathology and molecular subtypes of breast cancer. In order to explore other uses of HF ultrasound in real-time cancer diagnoses, two colorectal carcinoma cell lines were investigated: HT-29 and SW-620. This experiment tested the sensitivity of HF ultrasound on HT-29 and SW-620 and its ability to differentiate between the two lines. Subtypes of colorectal carcinoma have not been defined, but the morphology of these cell lines indicates that they may possess different molecular subtypes. Cell lines were grown as monolayer cultures and promptly tested with HF ultrasound using a single-element (50 MHz, 6.35-mm) ultrasonic immersion transducer. Ultrasound techniques and conditions were similar to previous breast cancer monolayer tests. Preliminary results indicate the cell lines produce different waveform and spectral signatures. The ability to differentiate between the two cell lines will both broaden the application of HF ultrasonic testing, and provide a method to define colorectal carcinoma subtypes. More cell lines will be tested in an effort to clearly define molecular subtypes of colorectal carcinoma. Defining subtypes will allow for more personalized diagnoses and treatment options for colon cancer patients.

9:15

3aBAa6. Acoustic power delivery for sub-millimeter dimension and deeply implanted medical devices. Marcus J. Weber, Jayant Charthad, Ting Chia Chang, and Amin Arbabian (Elec. Eng., Stanford Univ., 420 Via Palou Mall, Stanford, CA 94305, mjweber3@stanford.edu)

We propose to use acoustic power delivery for sub-mm^{3}medical implants, and we have designed acoustic receivers for investigating this technique of wireless power transmission. Compared with radio frequency and inductive transfer, acoustic power transfer gives more favorable impedance and available power for miniaturized and deeply implanted medical devices. Sub-mm^{3}dimensions are attractive for many different medical applications; however, there is little literature describing the properties or capabilities of miniaturized acoustic receivers, which is the topic of our investigation. Design of efficient miniaturized implants is challenging because of increased losses due to tissue absorption and coupling to parasitic resonant modes. We will present power transfer and impedance measurements of volumetrically scaled acoustic transducers through several thicknesses of tissue. Our measurements show significant available power with high output voltage, resulting from large transducer impedance, which is useful for overcoming threshold voltages of rectifier circuits. Preliminary measurements show the delivery of 340 μW of average AC power to a 1 mm^2 receiver through 3 cm of tissue with an intensity well below the FDA limit. In addition, we will compare our measurements with piezoelectric theory and discuss trends and limits of available power and impedance as a function of volume and transmit distance.

9:30

3aBAa7. Measurement of ultrasonic tissue characteristics of malignant colon cancer cells. Guðfríður Björg Möller, Madilena Mendiola (Phys., Mount Holyoke College, 50 College Ave., South Hadley, MA 01075, gudfridurb@gmail.com), Aislinn Daniels, Dalton Johnson, Rodoula Kyvelou-Kokkaliaris, Kenzi Watkins (Phys., Earlham College, Richmond, IN), and Maria-Teresa Herd (Phys., Mount Holyoke College, South Hadley, MA)

Colon cancer is the third most common cause of death by cancer resulting in over 50 000 deaths a year. The most common method for the detection of colon cancer, a colonoscopy, is quite invasive, and although non-invasive methods are available, they have not proven to be as effective. There have been successful studies on using ultrasound as a diagnostic tool for colon disease in rats. This suggests that ultrasound may be able to be a viable clinical tool for detecting colon cancer and gastrointestinal disease. Here, we explore using quantitative ultrasound to differentiate between benign and malignant colon cells. We made measurements of tissue characteristics of malignant colon cell pellets at different cellular densities. Using ultrasound frequencies ranging from 5 to 20 MHz, we measured the speed of sound, attenuation, and the backscatter coefficients (BSCs). Here, we present the results for the ultrasonic properties of cancerous colon cells and compare these for different densities of the cells.

9:45

3aBAa8. Cough monitoring for pulmonary tuberculosis using combined microphone/accelerometer measurements. Jingqi Fan (ECE, Tufts Univ., 161 College Ave., Medford, MA, jingqi.fan@tufts.edu), German Comina (Laboratorio de Ingeniería Física, Universidad Nacional de Ingeniería, Rimac, Peru), Robert Gilman (Dept. of Int.. Health, Johns Hopkins Bloomberg School of Public Health, Baltimore, MD), Jose Lopez (Unidad de Epidemiología, Hospital Nacional Dos de Mayo, Lima, Peru), and Brian H. Tracey (ECE, Tufts Univ., Medford, MA)

A laboratory-free test for assessing recovery from pulmonary tuberculosis (TB) would be very helpful in regions of the world where laboratory facilities are lacking. Our hypothesis is that analysis of cough sound recordings may provide such a test, as recovering patients should cough less frequently. We have carried out several studies on cough data from a cohort of TB patients in Lima, Peru [Larson *et al.*, PLOS One]. Our previous work provides a foundation to support larger-scale studies of coughing rates over time for TB patients undergoing treatment, but it only used recordings from lapel microphones. For the current study, we use an additional channel recorded by a throat-mounted accelerometer. The accelerometer only responds to patient-generated noise events and thus provides robustness to background noise in the environment. We describe algorithm development for cough data analysis using combined microphone/accelerometer measurements, and compare several event detection and classification strategies. We show that adding the accelerometer improves performance on detection and classification.

10:00

3aBAa9. Quantitative non-linear ultrasonic imaging of targets with high acoustic impedance contrast—Application to bone imaging. Régine Guillermin, Philippe Lasaygues, and Guy Rabau (Ondes et Imagerie, LMA/CNRS, 31, chemin Joseph Aigiuer, Marseille Cedex 20 13402, France, guillermin@lma.cnrs-mrs.fr)

This study focuses on the ultrasonic imaging of high impedance acoustical contrast targets. The aim is to obtain information about shape, dimensions, and sound speed profile of the studied objects. One domain of application is the characterization of long bones. Quantitative information about the acoustic properties of bones tissues are of great interest for diagnosing or treatment monitoring of bone diseases. Inverse scattering problems of this kind are non-linear and various approximations can be used to linearize the scattering equations. Classical methods based on the first-order Born approximation give good results for weakly scattering targets but fail when it comes to give a quantitative information especially for high impedance contrast targets such as bones. In the inversion algorithm proposed here, Green's theorem is used to obtain a domain integral representation of the scattered field. An iterative non-linear algorithm minimizing the discrepancy between the measured and computed scattered fields is used to reconstruct the sound speed profile in the region of interest. The minimization process is performed using a conjugated-gradient method. An experimental study was performed with targets made of paraffin and with lamb bones. Images of the sound speed profile obtained by inversion of experimental data are presented and discussed in both cases.

Session 3aBAb

Biomedical Acoustics: Biomedical Acoustics Best Paper Award Poster Session

Kevin J. Haworth, Chair
Univ. 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 $500 for first prize, $300 for second prize, and $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 their abstract numbers and titles listed. 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 10:30 a.m. to 12:00 noon.

laBAl0. Comparative analysis of small versus large transducers for high-frequency ultrasonic testing of breast cancer. Student author: **Madison J. Peterson**

laBA7. Improving ultrasound-based estimates of vector displacement fields in elastography applications. Student author: **Sanjay S. Yengul**

laBA8. Boundary conditions in quantitative elastic modulus imaging. Student author: **Daniel T. Seidl**

laBA9. Boundary condition-free elastic modulus reconstructions from ultrasound measured quasi-static displacements. Student author: **Olalekan A. Babaniyi**

lpBAlO. High-frequency ultrasonic measurement of vascularization in phantoms and Avastin-treated mice with breast tumors. Student author: **Andrea N. Quiroz**

lpBA6. Reproducibility of high-frequency ultrasonic signals in breast cancer detection. Student author: **A. Mackay Breivik**

lpBA7. Utah Valley University/Huntsman Cancer Institute Collaborative Breast Cancer Study: High-frequency ultrasound for margin assessments. Student author: **Andrew Chappell**

lpBA9. Ultrasonic phenotyping of breast cancer cells for molecular subtyping. Student author: **Laurel A. Thompson**

2aBA2. Passive mapping of acoustic sources within the human skull cavity with a hemispherical sparse array using computed tomography-based aberration corrections. Student author: **Ryan M. Jones**

3aBAa3. Treatment planning and strategies for acousto-optic guided high-intensity focused ultrasound therapies. Student author: **Matthew T. Adams**

3aBAa4. Using high-frequency ultrasound to detect cytoskeletal dysfunction in Alzheimer's disease. Student author: **Ashley N. Calder**

3aBAa5. Molecular subtyping of colorectal cancer using high-frequency ultrasound. Student author: **Alexis M. Holman**

3aBAa6. Acoustic power delivery for sub-millimeter dimension and deeply-implanted medical devices. Student author: **Marcus J. Weber**

3aBAa8. Cough monitoring for pulmonary tuberculosis using combined microphone/accelerometer measurements. Student author: **Jingqi Fan**

3pBAl. Effects of ambient pressure variation on the subharmonic response from contrast microbubbles. Student author **Nlma Mobadersany**

3pBA2. Ambient pressure estimation using subharmonic emissions from contrast microbubbles. Student author: **Krishna N. Kumar**

3pBA3. Acoustic characterization of polymer-encapsulated microbubbles with different shell-thickness-to-radius ratios using in vitro attenuation and scattering: Comparison between different rheological models. Student author: **Lang Xia**

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3pBA4. Nonlinear intravascular ultrasound contrast imaging with a modified clinical system. Student author: **Himanshu Shekhar**

3pBA6. StemBells: Localized stem cell delivery using targeted microbubbles and acoustic radiation force. Student author: **Tom Kokhuis**

3pBA8. Estimation of damping coefficient based on the impulse response of echogenic liposomes. Student author: **Jason L. Raymond**

3pBA9. The stable nonlinear acoustic response of free-floating lipid-coated microbubbles. Student author: **Ying Luan**

4pBAa2. Evaluation of sub-micron, ultrasound-responsive particles as a drug delivery strategy. Student author: **Rachel Myers**

4pBAa4. Response to ultrasound of two types of lipid-coated microbubbles observed with a high-speed optical camera. Student author: **Tom van Rooij**

4pBAa6. Acoustic levitation of gels: A proof-of-concept for thromboelastography. Student author: **Nate Gruver**

4pBAa7. Numerical simulations of ultrasound-lung interaction. Student author: **Brandon Patterson**

4pBAbl0. Surface roughness and air bubble effects on high-frequency ultrasonic measurements of tissue. Student author: **Percy D. Segura**

4pBAb4. Can quantitative synthetic aperture vascular elastography predict the stress distribution within the fibrous cap non-invasively. Student author: **Steven J. Huntzicker**

4pBAb5. Super wideband quantitative ultrasound imaging for trabecular bone with novel wideband single crystal transducer and frequency sweep measurement. Student author: **Liangjun Lin**

4pBAb8. Modeling ultrasonic scattering from high-concentration cell pellet biophantoms using polydisperse structure functions. Student author: **Aiguo Han**

4pBAb9. Characterizing collagen microstructure using high frequency ultrasound. Student author: **Karla P. Mercado**

WEDNESDAY MORNING, 7 MAY 2014 OMNI NARRAGANSETT A/B, 10:00 A.M. TO 12:00 NOON

Session 3aED

Hands-On Acoustics Demonstrations for Middle- and High-School Students

Andrew C. Morrison, Cochair
Natural Sci. Dept., Joliet Junior College, 1215 Houbolt Rd., Joliet, IL 60431

Cameron T. Vongsawad, Cochair
Phys. & Astronomy, Brigham Young Univ., 1041 E. Briar Ave., Provo, UT 84604

Acoustiacs has a long and rich history of physical demonstrations of fundamental (and not so fundamental) acoustics principles and phenomena. In this session “Hands-On” demonstrations will be set-up for a group of middle school students from the Providence area. The goal is to foster curiosity and excitement in science and acoustics at this critical stage in the students’ educational development and is part of the larger “Listen Up” education outreach effort by the ASA. Each station will be manned by an experienced acoustician who will help the students understand the principle being illustrated in each demo. Any acousticians wanting to participate in this fun event should email Andrew C. H. Morrison (amorrison@jjc.edu) or Cameron Vongsawad (cvongsawad@byu.edu).

Session 3aID

Interdisciplinary: Future of Acoustics

Paul D. Schomer, Chair
Schomer and Associates Inc., 2117 Robert Dr., Champaign, IL 61821

Invited Papers

11:00

3aID1. Criteria for the acoustic environment—What should we do? Paul D. Schomer (Schomer and Assoc. Inc., 2117 Robert Dr., Champaign, IL 61821, schomer@SchomerAndAssociates.com)

The questions are many; the answers I have are few, so I may call upon attendees of this session for help. Who should create policy for the acoustic environment? The F AA(aviation)? FHW A(roads)? DOE(energy)? DOD(defense)? or EPA? or DOI(interior/Park Service)? For example, it has been estimated that about 20 to 30% of the population are noise sensitive? Should policy be based on this minority or on the less sensitive majority? Currently, FAA and DOD allege that their criterion, DNL equal to 65 dB results in about 10 percent highly annoyed; the real percentage is 20 to 30%, so their criterion "protects" only the less sensitive. In a National Park, we are finding that hikers on a moderate length hike (about 4 to 10 km; 2 to 5 h) judge the pleasantness of the acoustic environment during the entire hike largely on the presence or absence of anthropogenic sound. Again there are factions to consider. Roughly 1/3 rate the overall pleasantness on the most pleasant portion of the hike, 1/6 rate on the least pleasant (most unpleasant), 1/6 on the average, and 1/3 on the most recent. What do we do? What should ASA do? Standards? WHY?

11:20

3aID2. Noise pollution in the 21st century. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1147, les@nonoise.org)

What will 21st century noise be and sound like? Will this century be noisier than the 20th century? What changes to noise policy and noise control are we likely? This paper explores these questions. The historical context of the last 100 years will be used to examine and understand the possibilities for the next 100 years.

11:40

3aID3. Future directions in psychoacoustic research facilities. Samuel Gordon, Roger EllingsonNCRAR, Veterans Affairs, 3710 SW US Veterans Hospital Rd., Portland, OR 97239, samuel.gordon@va.gov, David BrowningAltman Browning and Co., Portland, OR, Raymond JanischAcoust., Eckel Industries Inc., Cambridge, MA, and Frederick GallunNCRAR, Veterans Affairs, Portland, OR

The future of psychoacoustic research will involve the measurement of individual differences from a broader range of participants including those with physical disabilities. Auditory testing of physically disabled research participants in a fully anechoic environment presents challenges to the people being tested, to the research team, and to the facility. At foremost concern are the risks of personal injury to physically disabled persons while moving them into, and out from, the testing position in the anechoic chamber. Additional risks of injury exist to the researchers who assist with moving subjects into and out of the chamber. These risks need to be mitigated without compromising the acoustical performance of the anechoic chamber. This paper presents the requirements and the implemented design solutions for an 'Americans with Disabilities Act of 1990' compliant anechoic chamber that is currently being used for auditory research at the Department of Veterans Affairs, National Center for Rehabilitative Auditory Research facility in Portland, Oregon. We advocate that others consider accessibility and safety issues in the development of future psychoacoustic research facilities as we believe such factors are essential to the future of human testing and physical sciences.

Session 3aNS

Noise, Structural Acoustics and Vibration, and ASA Committee on Standards: Wind Turbine Noise

Nancy S. Timmerman, Cochair
Nancy S. Timmerman, P.E., 25 Upton St., Boston, MA 02118

Paul D. Schomer, Cochair
Schomer and Associates Inc., 2117 Robert Dr., Champaign, IL 61821

Kuangcheng Wu, Cochair
Newport News Shipbuilding, 202 Schembri Dr., Yorktown, VA 23693

Invited Papers

8:15

3aNS1. Public complaints about wind turbine noise and adverse health impacts justified. Stephen E. Ambrose (SE Ambrose & Assoc., 15 Great Falls Rd., Windham, ME 04062, seaa@myfairpoint.net), Robert W. Rand (Rand Acoustics, Boulder, CO), Richard R. James (E-Coustics Solutions, Okemos, MI), and Michael A. Nissenbaum (Medical Practice, Rort Kent, ME)

Significant proportions of IWT facility neighbors complain about turbine noise and sleep disturbances, among other adverse health complaints. We undertook an independent evaluation of several wind turbine projects located in Maine, Massachusetts, Vermont, New York, Illinois, Michigan, West Virginia, and Wisconsin to assess if common etiological factors exist. Adverse effects appear to relate to a basket of common factors that were overlooked or not included in preconstruction planning including noise predictions and assessments of likely community reactions. Correcting oversights in future projects should result in quieter IWT projects with reduced or no adverse community reactions. A unified methodology for doing so, enabling wind turbine developers, governmental agencies, municipal boards, and private citizens to assess for potential adverse noise impacts during the permitting phase is presented. Our results are consistent with prior USEPA studies, WHO assessments, and Pedersen and Waye research, among others.

8:35

3aNS2. Approximately 20 Hz plus harmonics amplitude modulated acoustic emissions from a 1.6 MW wind turbine, measurements versus predictions. Kevin A. Dooley (N/A, Toronto, ON, Canada) and Andy Metelka (N/A, 13652 Fourth Line, Acton, ON L7J 2L8, Canada, ametelka@cogeco.ca)

A recently presented hypothesis and model relating to the generation of spinning modes from wind turbines, as a direct result of acoustic interaction involving the tower, also predicts ~20 Hz plus harmonics low frequency amplitude modulated acoustic emissions as a side effect of the acoustic interaction. The low frequency sound is expected to propagate at measurable amplitudes to the far field (1 km to 2 km). Measurements focused on the ~20 Hz amplitude modulated fundamental and harmonics made at different angles relative to the rotor plane at very close range, and at greater distances are presented. The measurements are compared to predictions based on the tower acoustic interaction hypothesis.

8:55

3aNS3. Examination of the predictions versus measurements of an acoustic interaction model for spinning modes, and concurrent low frequency amplitude modulation acoustic signatures from a 3MW wind turbine. Kevin A. Dooley (N/A, N/A, 55-1817 Harbour Square, Toronto, ON M5J 2L1, Canada, kadooleyinc@rogers.com), Kristy Hansen, and Branko Zajamsek (Mech. Eng., Univ. of Adelaide, Adelaide, SA, Australia)

A recently presented hypothesis and model relating to the generation of spinning modes from wind turbines, as a direct result of acoustic interaction involving the tower, results in a far field infrasound sound pressure level prediction, which is higher than that predicted by point source method. The model also predicts a significant attenuation of the fundamental blade passing frequency component relative to the second and higher harmonics. The model concurrently predicts a low frequency (~20 Hz), amplitude modulated harmonic series as a side effect of the acoustic interaction on a 1.6 MW 80 m diameter wind turbine. This study examines the model predictions of a 3.0 MW 90 m diameter wind turbine, and compares the predictions to measurements of the low frequency harmonic series and blade passing frequency harmonics at several different distances from the wind turbine.

9:15

3aNS4. A comprehensive water tunnel test of a horizontal axis marine hydrokinetic turbine for model validation and verification. Arnie A. Fontaine, Ted G. Bagwell, Michael L. Johnson (Appl. Res. Lab., Penn State Univ., State College, PA), and Dean Capone (Appl. Res. Lab., Penn State Univ., PO Box 30, Sate College, PA 16803, dec5@psu.edu)

As interest in waterpower technologies has increased over the last few years, there has been a growing need for a public database of measured data for these devices. This would provide a basic understanding of the technology and means to validate analytic and numerical models. Through collaboration between Sandia National Laboratories, Penn State University Applied Research Laboratory, and University of California, Davis, a new marine hydrokinetic turbine rotor was designed, fabricated at 1:8.7-scale, and experimentally tested to provide an open platform and dataset for further study and development. The water tunnel test of this three-bladed, horizontal-axis rotor recorded power production, blade loading, near-wake characterization, cavitation effects, and noise generation. Additionally, preliminary comparisons are mode from unsteady CFD for the flow fields measured.

9:35

3aNS5. Equivalent sources method for supersonic intensity of arbitrarily shaped geometries. Nicolas P. Valdivia (Naval Res. Lab., 4555 Overlook Ave., SW, Washington, DC 20375, nicolas.valdivia@nrl.navy.mil)

Supersonic acoustic intensity is utilized to locate radiating regions on a complex vibrating structure. The supersonic intensity is obtained by a special process that removes the subsonic waves from the near-field acoustical holography measurement. The filtering process is well understood for separable geometries, but unfortunately, there are few results for arbitrarily shaped objects. This work proposes a methodology based on a stable invertible representation of the radiated power operator. The power operator is approximated numerically by the equivalent source formulation and the appropriate complete spectral basis is employed to form the stable invertible operator. The operator is formed with the most efficient radiation modes and these modes are utilized to obtain the supersonic solution for the near-field holographic problem. This concept is tested using numerically generated data in a spherical geometry and the results are validated with the spherical harmonic, supersonic filter. Finally, a vibrating ship-hull structure provides a physical example for application and validation of the proposed methodology in a more complex geometry. [This work was supported by the Office of Naval Research.]

Contributed Papers

9:55

3aNS6. Characterization of noise from an isolated intermediate-sized wind turbine. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

Community-based wind power companies provide subscriptions to individual homeowners and businesses for power generated by a locally installed turbine. Typically, such turbines are of an intermediate size, such as the Vestas V20 120-kW turbines operated by the Cascade Community Wind Company in several locations within Washington state. This model turbine has a tower height of 80 feet with a rotor diameter of 60 ft. Each turbine is installed individually on leased land, with no other turbines nearby. Noise measurements of a turbine located in Thorp, WA, were obtained in a variety of weather conditions. On several occasions with low to moderate wind speeds, the turbine was stopped, enabling the calculation of noise due to the turbine only. Results will be presented showing spectral content and sound pressure level contours for a range of wind speeds.

10:10

3aNS7. Investigations on psychoacoustical and non-acoustical moderators for annoyance evoked by wind turbine noise. Leonid Schmidt (Audio Commun. group, Tech. Univ. Berlin, Blüch4erplatz 14, Aachen 52068, Germany, leonid.schmidt@gmx.net) and André Fiebig (HEAD Acoustics, Herzogenrath, Germany)

In 2012, in total around 23 000 wind turbines are installed on land in Germany. These wind turbines are important to achieve a change in Germany's current energy policy gaining more energy from renewable resources. However, current research underlines that the noise of wind turbines causes noise annoyance and provoke complaints. The sound characteristics of wind turbines depend on many different variables, e.g., type of wind turbine and wind speed. Unfortunately, only little is known about the specific noise characteristics, which are mainly responsible for noise annoyance. Therefore, laboratory experiments are carried out to identify the most annoying noise characteristics of wind turbine noise. The laboratory experiment includes an evaluation of different sounds from wind turbines, manipulated wind turbine sounds, and sounds from other noise sources. The work intends to improve the understanding about the role of psychoacoustic parameters going beyond equivalent continuous sound level. Additionally, the relevance of non-acoustical factors for annoyance caused by wind turbine noise is investigated by interviewing extensively the test subjects.

Session 3aPA

Physical Acoustics: Acoustical Methods and Sensors for Challenging Environments

Dipen N. Sinha, Cochair
Sensors & Electrochemical Devices, Los Alamos Natl. Lab., MPA-11 D429, PO Box 1663, Los Alamos, NM 87545

Cristian Pantea, Cochair
MPA-11, Los Alamos Natl. Lab., MS D429, Los Alamos, NM 87545

Chair's Introduction—7:55

Invited Papers

8:00

3aPA1. Acoustic measurements of rock formations in oilfield boreholes. David L. Johnson (Schlumberger-Doll Res., One Hampshire St., Cambridge, MA 02139, johnson10@slb.com)

Once a borehole is drilled into a rock formation as a potential oil or gas well the environment needs to be characterized by a variety of physical measurements so that e.g. one may know at what depths the hydrocarbon, if any, is located. In this talk I will outline some techniques for measuring, *in situ*, the compressional and shear speeds of sound in a rock formation as a function of depth. Here the "difficult and challenging conditions" are that the measuring instrument is inside the sample (rather than the other way around), the temperatures may reach 175 °C, and the pressure in the borehole may reach as high as 1000 atmospheres.

8:20

3aPA2. Acceleration of acoustical emission precursors preceding failure in sheared granular material. Paul A. Johnson (Geophys., LANL, MS D443, Los Alamos, NM 87545, paj@lanl.gov)

Earthquake precursor observations are becoming progressively more widespread as instrumentation improves, in particular, for interplate earthquakes (e.g., Bouchon *et al*., Nature Geosci., 2013). One question regarding precursor behavior is whether or not they are due to a triggering cascade where one precursor triggers the next, or if they are independent events resulting from slow slip. We investigate this topic in order to characterize the physics of precursors, by applying laboratory experiments of sheared granular media in a bi-axial configuration. We sheared layers of glass beads under applied normal loads of 2–8 MPa, shearing rates of 5–10 μm/s at room temperature and humidity. We show that above ~3 MPa load, precursors are manifest by an exponential increase in time of the acoustic emission (AE), with an additional acceleration of event rate leading to the primary stick-slip failure event. The recorded AE are clearly correlated with small drops in shear stress during slow slip prior to the main stick-slip failure. Event precursors take place where the material is still modestly dilating, yet while the macroscopic frictional strength is no longer increasing. The precursors are of order 100$\times$ smaller in recorded strain amplitude than the stick-slip events. We are currently working on statistical methods to determine whether or not the precursors are triggered cascades. [Bouchon *et al*., Nature Geosci. **6**, 299–302 (2013).]

8:40

3aPA3. Using nonlinear ultrasound to measure microstructural changes due to radiation damage in steel. Laurence Jacobs, Kathryn Matlack, Jin-Yeon Kim (Mech. Eng., Georgia Tech, COE Georgia Tech, 225 North Ave. Tech Tower, Atlanta, GA 30332-0360, laurence.jacobs@coe.gatech.edu), Jianmin Qu (civil Eng., Northwestern Univ., Evanston, IL), and Wall J. Joe (EPRI, Charlotte, NC)

The planned life extension of nuclear reactors throughout the United States and abroad will cause reactor vessel and internals materials to be exposed to more neutron irradiation than was originally intended. A nondestructive evaluation (NDE) method to monitor radiation damage would enable safe and cost-effective continued operation of nuclear reactors. Nonlinear ultrasound is an NDE technique that is sensitive to microstructural changes in metallic materials, such as dislocations, precipitates, and their combinations, which are quantified by the measurable acoustic nonlinearity parameter. Recent research has shown the sensitivity of the acoustic nonlinearity parameter to increasing neutron fluence in representative Reactor Pressure Vessel (RPV) steels. The current work considers nonlinear ultrasonic experiments conducted on similar RPV steel samples that had a combination of irradiation, annealing, re-irradiation, and/or re-annealing to a total neutron fluence of 0.5–5 x 1019 n/cm^2(E > 1 MeV) at an irradiation temperature of 290°C. The acoustic nonlinearity parameter generally increased with increasing neutron fluence, and consistently decreased from the irradiated to the annealed state over different levels of neutron fluence. This comprehensive set of results illustrates the dependence of the measured acoustic nonlinearity parameter on neutron fluence, material composition, irradiation temperature, and annealing.

9:00

3aPA4. Materials and fabrication techniques for resonant ultrasound spectroscopy at high and low temperatures. Albert Migliori (NSEC-NHMFL, Los Alamos Natl. Lab., MS E536 Los Alamos, NM 87545, migliori@lanl.gov)

Measurement of the mechanical resonances of materials of interest to condensed matter science is becoming increasingly common because it reveals important thermodynamic signatures such as phase transitions, as well as providing sound speeds and stiffness information for technology. Often done using Resonant Ultrasound Spectroscopy, the ultimate precision of measurements is determined by the mechanical Q, not unusually 104 or higher, thereby making it possible to determine changes in elastic moduli at the sub pert-per-million level. However, resonances and changes in the acoustic response of the cell that holds transducers and the specimen to be measured can introduce artifacts as temperatures change, clouding otherwise important observations. We describe here solutions to such problems with acoustically "dead" materials capable of operation from below 1 K to 900 K using easily available starting components. We also describe strategies for electrical contacts at temperature above the melting point of lead-tin solder. Some unusual results are presented.

9:20

3aPA5. Harsh environment sensors: Aircraft and oil field applications of electromagnetic and acoustic technologies. Edward R. Furlong (General Electric, 1100 Technol. Park Dr., Billerica, MA 01821, ted.furlong@ge.com)

There is an old saying that what can be measured can be improved. However, measuring the key parameters in aircraft engines that drive efficiency and emissions is very difficult. This is even more the case in oil field applications, where extremely high pressures and temperatures are commonly encountered. But the benefits to society of reduced emissions (carbon, pollutants, and noise) and improved oil and gas recovery are tremendous. The task for instrument manufacturers is to develop sensors that are both reliable and cost effective. Recent advances in the application of electromagnetic and acoustic technologies are described for measuring temperature, pressure, flow, and composition in harsh environments. The sensors range from inductively and optically coupled ceramic and silicon devices to ultrasonic and SAW devices packaged with high temperature electronics to fiber optic systems. These sensors are now being deployed on new airframes, engines, deep water wells, horizontal tight shale formations, and high temperature/high pressure wells.

9:40

3aPA6. A resonance technique for the acoustic characterization of liquids in harsh environments. Blake Sturtevant (Los Alamos National Lab., PO Box 1663, MS D429, Los Alamos, NM 87545, bsturtev@lanl.gov)

Accurate knowledge of a liquid's acoustic properties, such as sound speed as a function of temperature and pressure, is important for both basic and applied science. For basic science, sound speed is important for constraining thermodynamic equations-of-state as well as determining elastic nonlinearity parameters. From an applied perspective, sound speed can be used together with other properties to monitor fluid temperature, pressure, and composition. There are important applications, such as in oil/gas or geothermal well characterization, where it is desirable to measure sound speed in liquids in well bore under high pressure, high temperature, and in corrosive environments. However, few experimental sound speeds have been previously reported above 100°C even in liquids as common as water. This talk focuses on the development of a portable, rugged, resonance-based measurement cell for high precision (better than 0.1%) *in-situ* measurements of sound speed in high temperature, high pressure, and corrosive liquids. As an example of the technique, experimentally determined sound speeds in liquid water up to 250°C and 3000 psi will be presented. Acoustic nonlinearity in water, as determined from sound speed as a function of temperature and pressure, will also be discussed.

10:00–10:15 Break

Contributed Papers

10:15

3aPA7. Development, qualification, and performance validation of an optical differential pressure sensor for downhole permanent monitoring applications. James R. Dunphy (Reservoir Monitoring, Weatherford, Wallingford, CT), Omer H. Unalmis (In-Well Flow, Weatherford, 22001 North Park Dr., Kingwood, TX 77339, haldun.unalmis@weatherford.com), and Domino Taverner (Reservoir Monitoring, Weatherford, Wallingford, CT)

This paper describes the development, qualification, and performance validation of an optical differential pressure (DP) sensor for the high-pressure and high-temperature downhole environment. The current implementation of measuring DP in downhole applications is based on the calculated difference between two static pressure sensor measurements with high uncertainties. A true DP sensor is superior in comparison due to the decreased uncertainty both in measurement and component levels. This translates to better performance in system level measurements such as flow. A development program was launched in 2007 to build an optical DP sensor targeted for two main applications: standalone use in single-phase or auxiliary use in multiphase flow measurement systems. Several prototypes have gone through multiple design phases and qualification test programs including mechanical shock and vibration tests, thermal tests, hot vibration tests, short-term stability and long-term endurance tests. The sensor was then integrated with a Venturi and tested in a single-phase flow loop to validate its performance against an electronic DP sensor and an electromagnetic flowmeter. Results suggest that optical DP sensor performs better than the electronic DP sensor in measuring differential pressures, and is on a par with the electromagnetic flowmeter.

10:30

3aPA8. Thermoacoustic engines as self-powered sensors within a nuclear reactor. Steven L. Garrett (Grad. Prog. in Acoust., Penn State, Appl. Res. Lab, P. O. Box 30, State College, PA 16804-0030, sxg185@psu.edu), Randall A. Ali (Elec. Eng., Univ. of the West Indies, Port-of-Spain, Trinidad and Tobago), and James A. Smith (Fundamental Fuel Properties, Idaho National Lab., Idaho Falls, ID)

The core of a nuclear reactor is a particularly harsh environment when functioning properly. When there is an "incident," possibly with the loss of electrical service that accompanied the earthquake and tsunami that struck the Fukushimi Daiichi reactors on 11 March 2011, the term "harsh" seems too tame. We review the development and testing of very simple standing--wave thermoacoustic engines that can be configured as nuclear fuel rods to exploit the temperature differences within that environment rather than try to shield the sensor from "harshness" [U.S. Pat. Appl. Serial No. 13/968,936 (Aug. 16, 2013)]. Those engines produce high amplitude sound that couples to the surrounding heat-transfer fluid to telemeter the information (as frequency and amplitude) to the exterior of the reactor vessel, again without requiring external electrical power. Laboratory results will demonstrate measurement of coolant temperature, identify evolved gases, and provide information about changes in porosity of solids. Thermoacoustic resonances are maintained without use of either an explicit physical hot or cold heat exchanger—a ceramic "stack" is the only required component. We suggest extensions of this approach to other processes that generate substantial temperature gradients, such as industrial crucibles for melting glasses and metals. [Work supported by the U.S. Department of Energy.]

10:45

3aPA9. Nuclear material identification using resonant ultrasound spectroscopy. Cristian Pantea, Tarik A. Saleh, Albert Migliori, Jonathan B. Betts, Erik P. Luther, and Darrin B. Byler (Los Alamos National Lab., MS D429, Los Alamos, NM 87545, pantea@lanl.gov)

Resonant ultrasound spectroscopy (RUS) is a well-established method for determination of the full tensor of elastic moduli of a solid sample in a single frequency sweep. The elastic moduli, together with density, can provide information related to materials fabrication processes, providing a unique signature, or fingerprint, of a material. The goal of this study was to provide forensics for nuclear materials in solid ceramic or metallic form, including composition. The premise of this study was that it is really difficult to find two materials whose density and shear and /or bulk modulus match. We used RUS to determine the bulk and shear modulus for a total of 27 samples. The samples consisted of depleted uranium oxide (MOX) with different doping of Ce, Pu, and Nd oxides, and different methods of fabrication. They were in form of cylinders with flat and parallel faces. Two-dimensional and three-dimensional spaces were investigated, using shear modulus, bulk modulus, and density as variables. The densities varied between 9.0 and 10.6 g/cc, while the shear modulus was 55–80 GPa, with a bulk modulus of 150–240 GPa. The results obtained suggest that there is a good correlation between the elastic moduli and density for samples of different compositions/origins.

11:00

3aPA10. Resonance ultrasound spectroscopy measurements of sandstone at high temperature. Eric S. Davis, Blake T. Sturtevant, Dipen N. Sinha, and Cristian Pantea (Los Alamos Natl. Lab., MPA-11, MS D429, Los Alamos, NM 87545, e.s.davis@tcu.edu)

Deep underground wells, such as those of interest to the oil and gas as well as geothermal industries, are often found in large sandstone formations. In order for drilling, enhancement, and advanced engineering techniques such as hydraulic fracturing to be efficient and successful, the mechanical properties of materials that make up the reservoir must be accurately known. We have used resonant ultrasound spectroscopy (RUS) to determine the physical properties of Berea Sandstone, such as the elastic moduli. In contrast to single crystals or high quality polycrystalline samples, the porous and attenuating nature of sandstone makes an acoustic study of sandstone very challenging. Additionally, the sandstones must be studied at high temperatures in order to simulate conditions that are found in the field. We will present our work on the temperature dependence of the elastic moduli of sandstone (between room temperature and 205 °C.) Our measurements show that Berea sandstone is a very soft material with a bulk modulus of about 6 GPa as compared to 76 GPa for aluminum. Furthermore, a ~10% softening was observed with decrease in temperature, down to a temperature of 110°C, followed by a ~7% hardening down to ambient temperature.

11:15

3aPA11. The propagation of sound above and within a hardbacked rigid porous layer. Hongdan Tao, Bao N. Tong, and Kai Ming Li (Mech. Eng., Purdue Univ., 140 South Martin Jischke, West Lafayette, IN 47907-2031, mmkmli@purdue.edu)

The present paper examines, theoretically and experimentally, the sound field in the vicinity of a non-locally medium due to an airborne source. The non-locally reacting medium is characterized by a porous layer of finite thickness which is placed on a perfectly reflecting plane. According to an asymptotic analysis, the total sound field within the rigid porous medium consists of two components. Each of these two components can be represented by an integral expression. They can then be evaluated by a standard saddle path method to obtain a uniform asymptotic solution. Numerical validation with wave-based numerical schemes demonstrates the accuracy and computational efficiency of the derived asymptotic formula. Additional validation is provided through indoor experimental data obtained by using a layer of glass beads for modeling the rigid porous medium. When the receiver is situated within the porous layer near the perfectly reflecting plane, experimental measurements and theoretical predictions suggest that the interaction of the refracted wave with the perfectly reflecting plane has a significant impact on the total sound field. Experimental data and numerical simulations also indicate that it is rather difficult to distinguish the results between a thin rigid porous layer and a semi-infinite rigid porous medium for an airborne receiver.

Session 3aPPa

Psychological and Physiological Acoustics: Diagnostics of the Pathological Middle Ear by Wideband Acoustic Impedance/Reflectance Measures

Jont B. Allen, Chair
2061 Beckman Inst, 405. N. Mathews, Urbana, IL 61801

Chair's Introduction—8:00

Invited Papers

8:05

3aPPa1. Non-invasive methods for diagnosing diseases of the ear: Wideband acoustic immittance and umbo velocity. Gabrielle R. Merchant (Speech & Hearing BioSci. & Technol., Eaton-Peabody Lab., Harvard-MIT Div. of Health Sci. and Technol., Massachusetts Eye and Ear Infirmary, EPL, Massachusetts Eye & Ear Infirmary, 243 Charles St., Boston, MA 02114, gmerchan@MIT.EDU), John J. Rosowski and Hideko H. Nakajima (Dept. Otology & Laryngology, Eaton-Peabody Lab., Harvard Med. School, Massachusetts Eye and Ear Infirmary, Boston, MA)

Measurements of ossicular motion using laser Doppler vibrometry (LDV) in patients with various ear diseases, along with fresh cadaveric experiments, have increased our understanding of how sound is transduced to the cochlea and how this transduction is modified by various conductive pathologies. These studies have dispelled some misguided beliefs, changed clinical treatments, and shown the potential of LDV as a diagnostic. LDV, however, has substantial limitations as a clinical tool. For patients with conductive hearing loss of unknown etiology (where general otologic exam and conventional tympanometry are not diagnostic), another non-invasive measurement, wideband acoustic immittance (WAI, directly related to power reflectance), in conjunction with audiometric measurements, can differentiate between ossicular fixation, ossicular discontinuity and superior canal dehiscence (SCD). Furthermore, WAI measurements show a common pattern in power reflectance in patients with SCD, with or without a conductive hearing loss. An algorithm to identify this pattern in power reflectance suggests that WAI may be a simple, inexpensive screening tool for SCD. Evidence will be presented from our studies on patients with various otologic diseases, as well as from fresh cadaveric preparations simulating various diseases. Power reflectance can assist in treatment decisions and prevent unnecessary and inappropriate treatments and surgeries.

8:25

3aPPa2. Identifying otosclerosis with battery of aural acoustical tests of absorbance, group delay, reflex threshold, and chirp-evoked otoacoustic emissions. Douglas H. Keefe, Kelly L. Archer (Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68131, Douglas.Keefe@boystown.org), Kendra K. Schmid (Dept. of Biostatistics, College of Public Health Masters Programs, Nebraska Medical College, Omaha, Oregon), Denis F. Fitzpatrick (Boys Town Natl. Res. Hospital, Omaha, NE), M. Patrick Feeney (Natl. Ctr. for Rehabilitative Auditory Res., Veterans Administration and Oregon Health & Sci. Univ., Portland, OR), and Lisa L. Hunter (Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH)

This study evaluated the clinical utility in diagnosing otosclerosis with aural acoustical tests of absorbance, acoustic reflex threshold (ART), and otoacoustic emissions (OAEs) in 23 normal-hearing (NH) ears, 12 ears diagnosed with otosclerosis (OS), and 13 ears after surgical intervention (SU) for otosclerosis. Subjects received audiometric evaluations, and tests of ipsilateral/contralateral ART, pressure reflectance (0.25–8 kHz) parameterized by absorbance and group delay at ambient pressure and at swept tympanometric pressures, and chirp-evoked OAEs (1–8 kHz). ARTs were measured using tonal and broadband noise activators, based on differences in wideband absorbed sound power before and after activator presentation. For NH compared to OS ears, mean ambient absorbance was larger at 4 kHz; mean tympanometric absorbance had larger peak-to-tail differences at low and high frequencies. Probe-to-eardrum length was estimated using group delay at the frequency of the minimum absorbance above 2 kHz, and combined with acoustically estimated area to calculate wideband compensated admittance at the eardrum. Absorbance and compensated group delay revealed complementary information on middle-ear function. Typical OS and SU ear tests showed absent TEOAEs and ARTs, reduced absorbance in OS ears, and anomalous reflectance <1 kHz in SU ears. Other middle-ear conditions showed different patterns of test-battery responses. [Research supported by NIH.]

8:45

3aPPa3. Identification of conductive hearing loss in infants using maximum likelihood analysis of wideband acoustic absorbance and admittance. Beth Prieve and Hammam AlMakadma (Commun. Sci. and Disord., Syracuse Univ., 621 Skytop Rd., Syracuse, NY 13244, baprieve@syr.edu)

Wideband acoustic absorbance (WAA)/reflectance measures of the middle ear identify conductive hearing loss (CHL) in infants and children with excellent accuracy. Recent literature has indicated that WAA analyzed using maximum likelihood ratios in children with conductive hearing loss were more accurate than the clinical standard of single-frequency tympanometry at one frequency. In infants,

single-frequency tympanometry is as effective in identifying conductive hearing loss as wideband acoustic reflectance in one frequency band. The question that arises is whether identification of conductive hearing loss in infants is different than that from children, or, if multiple variable analysis of WAA contributes to the highly significant outcomes. The current project used maximum likelihood ratios to analyze both WAA and admittance measured through tympanometry using three probe tone frequencies. WAA and tympanometry identified CHL equally well in infants, and WAA results are similar to those reported for children. The results suggest that including several frequencies to measure conductive properties of the outer and middle ear is more powerful than single frequencies or bands.

9:05

3aPPa4. Comparisons of reflectance measurements across measurements sessions, instruments, and ages. Susan E. Voss, Defne Abur, Hiwot Kassaye (Eng., Smith College, 100 Green St., Northampton, MA 01063, svoss@smith.edu), and Nicholas J. Horton (Mathematics, Amherst College, Amherst, MA)

Wideband acoustic immitance measures (WAI) are an active area of research aimed at the development of an objective and noninvasive audiometric test that can reliably identify a range of middle-ear disorders. This work compares repeated WAI measurements (absorbance and its closely related quantity power reflectance, in particular) made on normal hearing subjects. In particular, measurements were made on the left and right ears of nine subjects ages 19–22 and seven subjects ages 41 to 47. Each subject returned for repeated measurement sessions between four and eight times. At each measurement session, WAI was measured using two distinct FDA-approved devices: HearID from Mimosa Acoustics and Titan from Interacoustics. This presentation will compare the WAI measurements between two distinct age groups, the two distinct instruments, and across measurement sessions. Additional analyses will be presented to determine if assumptions about ear-canal areas might explain differences between the systems from Mimosa Acoustics and Interacoustics.

9:25

3aPPa5. Estimating the residual ear canal contribution to complex acoustic reflectance measurements using pole-zero fitting. Sarah Robinson (Elec. Eng., Univ. of Illinois at Urbana-Champaign, 2137 Beckman Inst., MC 251, 405 N Mathews Ave., Urbana, IL 61801, srrobin2@illinois.edu), Suzanne Thompson (Commun. Sci. and Disord., St. John's Univ., Queens, NY), and Jont Allen (Elec. Eng., Univ. of Illinois at Urbana-Champaign, Urbana, IL)

For diagnostic acoustic measurements of the middle ear, the residual ear canal (REC) between the probe and tympanic membrane (TM) is a significant source of non-pathological variability. Tympanometry measures the TM compliance as a function of canal static pressure, at a single frequency (226 Hz). Alternatively, wideband reflectance is measured at ambient pressure, over a large frequency range (0.2–6.0 kHz). To account for the REC effect, tympanometry assumes that the compliance tends to zero at large static pressures, which may not be a valid assumption (Rabinowitz, 1981), whereas reflectance assumes that the REC contributes a lossless delay. Previously, the authors developed a method to parameterize complex reflectance measurements using pole-zero fits, which may be factored into all-pass and minimum-phase components. The lossless all-pass component approximates the unknown REC delay, while the low-frequency TM compliance may be estimated from the minimum-phase component. Using this approach, we evaluate middle ear static pressure data from a controlled study of cadaver ears (Voss, 2012) and an *in vivo* study in which subjects were trained to induce negative middle ear pressure, as well as controlled syringe measurements. Our results indicate that the TM compliance is not zero at the static pressure extremes measured under tympanometry.

9:45

3aPPa6. Development of a finite element model for normal and pathological middle ears: Impedance, reflectance, and sweep frequency impedance. Sunil Puria, Hongxue Cai (Mech. Eng., Stanford Univ., 496 Lomita Mall, Stanford, CA 94305, puria@stanford.edu), Shinji Hamanishi (Mech. Eng., Sendai National College of Technol., Miyagi, Japan), Kevin N. O'Connor, and Charles Steele (Mech. Eng., Stanford Univ., Stanford, CA)

A 3D 'virtual middle ear model' using finite-element modeling techniques was developed to simulate the dynamics of the human middle ear. COMSOLMultiphysics software was used to solve the resulting acoustics-structure interaction problem. The model is validated by comparing numerical results with experimental data measured in the ear canal (EC), on the tympanic membrane (TM), umbo, stapes footplate, and cochlear pressure. The model consists of anatomy from μCT imaging and material parameters from the literature (Cai *et al*., PLOS One, in review). The EC impedance Zec, reflectance Rec, and the pressure Pec due to a constant displacement Dec (Wada *et al*., 1998) were calculated for the normal middle ear, with the stapes blocked, and with the stapes disarticulated. The results in this virtual model are consistent with experiments performed in both cadaveric and living ears. The model is used to analyze the sensitivity and specificity of Zec, Rec, and Pec due to variations in the material properties of the middle ear including the TM, ossicles, malleus-incus, and incus-stapes joints, and the footplate. [Work supported in part by grant R01-DC05960 from the NIDCD of NIH and by a Fellowship from the Institute of National Colleges of Technology, Japan.]

Contributed Papers

10:05

3aPPa7. Cochlear reflectance: Measurements and modeling. Daniel Rasetshwane and Stephen T. Neely (Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68131, daniel.rasetshwane@boystown.org)

Cochlear reflectance (CR), the cochlear contribution to ear-canal reflectance (ECR), has theoretical advantages for cochlear modeling. Comparisons between measurements and models may lead to improved interpretation of cochlear status and provide a basis for making improvements to the models. Previous evaluation of clinical utility of CR measurements showed that CR did not predict auditory status or behavioral threshold as accurately as other otoacoustic emission measurements. Strategies for improving the quality of CR measurements were assessed in ECR measurements from 27 participants. Results indicate that the quality of CR measurements can be improved by (1) increased averaging time and (2) adjustment to the methods for extracting CR from ECR. Simulation of ECR was performed using a combination of a middle-ear model and a one-dimensional cochlear model. Simulated CR was the ECR difference between

active and passive conditions of the model. The model simulation results were compared with measurements of ECR and CR in both the time and frequency domains. Minor disparities between measurements and model will provide a basis for improvements in the model. Substantial agreement between measurements and model suggest that CR is consistent with linear coherent reflection due to random impedance perturbations along the cochlear partition.

10:20

3aPPa8. Acoustic horn reflectance: Equations and measurements. Stephen T. Neely and Daniel Rasetshwane (Boys Town Natl. Res. Hospital, 555 North 30th St., Omaha, NE 68131, stephen.neely@boystown.org)

Reflectance is the transfer function between forward and reflected components of pressure waves that propagate in wave guides such as acoustic horns. Exact solutions to Webster's Horn Equation are only known for a few specific shapes, including parabolic, conical, and exponential. Explicit equations for reflectance in these three horn shapes were recently published for infinite-length horns. Measured reflectance in 3D-printed, finite-length examples of these horn shapes, show no similarity in the frequency-domain to exact reflectance for infinite-length horns. The similarity improves after adjustments to both the equations and the measurements. New equations were derived for exact reflectance of finite-length horns. Measured reflectance was smoothed by time-domain windowing. In contrast to frequency-domain reflectance, comparisons of time-domain reflectance prior to the time sound reaches the end of the horn were not much affected by these adjustments. Because exact equations are known and 3D-printed examples are easy to obtain, these three horn shapes may be useful as standards for comparing different reflectance-measurement systems.

WEDNESDAY MORNING, 7 MAY 2014 BALLROOM A, 8:00 A.M. TO 11:00 A.M.

Session 3aPPb

Psychological and Physiological Acoustics: Binaural Processing and Spatial Perception (Poster Session)

Jayaganesh Swaminathan, Chair
Boston Univ., 635 Commonwealth Ave., Rm. 320, Boston, MA 02215

All posters will be on display from 8:00 a.m. to 11:00 a.m. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 9.30 a.m. and contributors of even-numbered papers will be at their posters from 9:30 a.m. to 11:00 a.m.

Contributed Papers

3aPPb1. Investigating stream segregation and spatial hearing using event-related brain responses. Le Wang (CompNet, Boston Univ., 677 Beacon St., Boston, MA 02215, lwang@bu.edu), Samantha Messier (Biomedical Eng., Boston Univ., Boston, MA), Scott Bressler (CompNet, Boston Univ., Boston, MA), Elyse Sussman (Neurosci., Albert Einstein College of Medicine , Bronx, NY), and Barbara Shinn-Cunningham (CompNet, Boston Univ., Boston, MA)

Several studies have used auditory mismatch negativity (MMN) to study auditory stream segregation. Few of these studies, however, focused on the stream segregation that involves spatial hearing. The present study used MMNs to examine the spatial aspect of stream segregation. Traditional oddball streams were presented in a passive listening paradigm, either in isolation or in the presence of an interfering stream. The interfering streams were engineered so that the deviants were not unexpected if the two streams were heard as perceptually integrated. Interfering streams were either spectrally distant from or close to the oddball stream, and were also spatially separated from the oddball stream. The deviant stimuli differed from the standards in perceived spatial location. For comparison, the MMN paradigm developed by Lepistö *et al.* (2009) using intensity deviants was repeated on the same group of subjects. For both paradigms, the MMN was strongest when the oddball stream was presented in isolation, less strong but present when the two streams were spectrally separated, and not observable when the streams were spectrally close. These results demonstrate the feasibility of using the MMN to measure spatial stream segregation, especially in populations for whom task-based behavioral experiments cannot be undertaken.

3aPPb2. Spatial influences on change detection within complex auditory scenes. Kelly Dickerson (Army Res. Lab., 131 Waldon Rd., Abingdon, MD 21009, dickersonkelly23@gmail.com), Jeremy Gaston (Army Res. Lab., Aberdeen, Massachusetts), and Angelique Scharine (Army Res. Lab., Aberdeen, MD)

Change deafness is the auditory analog to change blindness. Both phenomena represent a tendency to miss large changes in the environment, suggesting that sensory experiences are not verbatim and may details crucial for detection and identification. Spatial separation facilitates change detection in vision, but its role in auditory change detection is unclear. In this study, we examined the impact of spatial separation on the detection of appearing and or disappearing sound sources in an auditory scene. Participants listened to a brief auditory scene (1000 ms) comprised of four sources followed by a scene where a sound source was added or subtracted from the scene or where no change occurred. There were two listening conditions, where the sound sources were each distributed across a loudspeaker array, or the sound sources were all played from a single loudspeaker. Results indicate that listeners were better able to detect appearing than disappearing sounds, and fewer errors were made when sound sources were spatially separated. These results are consistent with an attention-based explanation of change detection failures. Further, the beneficial effect of spatial separation suggests that change blindness and deafness may share a common mechanism.

3aPPb3. Spatial attention in an auditory dual task. Nandini Iyer (Air Force Res. Lab., 2610 Seventh St., Bldg. 441, Area B, Wright Patterson Air Force Base, OH 45433, Nandini.Iyer@wpafb.af.mil), Eric R. Thompson (Ball Aerosp., Dayton, OH), Griffin D. Romigh (Air Force Res. Lab., Dayton, OH), Carryl L. Baldwin (George Mason Univ., Fairfax, VA), and Brian D. Simpson (Air Force Res. Lab., Dayton, OH)

Research on divided auditory attention has focused on the ability of listeners to report keywords from two spatially separated simultaneous talkers (Best *et al*., 2006); however, the information that listeners extract from each talker is the same (i.e., keyword identification). In realistic listening environments, there is often a competing demand for auditory attention; listeners might be required to monitor critical auditory events while also responding to an ongoing auditory signal. It is not clear how spatial separation affects performance in auditory dual-tasks under varying levels of task difficulty. Listeners identified an ongoing stream of color/number keywords originating at 0° azimuth (primary task), while detecting the presence of a critical call sign originating from locations ranging from −45° to +45° (secondary task). The difficulty of the primary task was varied by introducing noise or by requiring stimuli identification in an auditory one- or two-back memory recall task. The difficulty of the secondary task was varied by increasing the set-size of critical call signs that listeners had to monitor and by changing the SNRs. Preliminary results indicate that a listener's ability to detect the presence of a critical call sign increased as its spatial location moved further away from 0° azimuth and performance was modulated by the difficulty of the primary task. The result has important implications for spatial attention as a function of task difficulty in multitalker environments.

3aPPb4. Mislateralization of sound images at dichotically presented a rippled noise signal and masker. Olga Milekhina and Alexander Supin (Inst. of Ecology and Evolution, 33 Leninsky Prospect, Moscow 119071, Russian Federation, alex_supin@mail.ru)

A mislateralization effect was observed when a target signal and a masker were presented dichotically: the target signal in one ear and the masker in the other ear. The target signal was band-filtered rippled noise with periodical interchange of ripple peaks and troughs positions. This interchange produced periodical timbre variations in the sound image. The masker was non-rippled band-filtered noise. In 100% of the trials, the image of periodical variations of timbre reflecting interchanges of the ripple peaks and troughs in the target signal was perceived (released from masking) at low ripple density and signal/masker ratios down to -35 dB. However, lateralization of the image depended on the signal/masker ratio. The image was correctly lateralized toward the ear of the target signal when the signal was of a higher level than the masker. The image was wrongly lateralized toward the ear of the masker presentation (mislateralized) when the masker was of a higher level than the signal. Implications to mechanisms of dichotic release from masking are discussed: the release is possible without the spatial-attention mechanism.

3aPPb5. Effects of cognitive load on selective and divided auditory spatial attention. Daniel McCloy (Inst. for Learning and Brain Sci., Univ. of Washington, Box 357988, Seattle, WA 98115-7988, drmccloy@uw.edu) and Adrian KC Lee (Speech and Hearing Sci., Univ. of Washington, Seattle, WA)

In a previous experiment [McCloy and Lee, 2013, *J. Acoust. Soc. Am.* **134**, 4230 (2013)], we reported an asymmetry between phonetic and semantic detection tasks with respect to spatially separated vs. spatially adjacent attended word streams: phonetic tasks showed high false alarm rates when to-be-attended streams were spatially divided, while semantic tasks did not. In this experiment, we manipulate the difficulty of the semantic task to investigate the effect of cognitive load on task performance. We compare trials in which the two to-be-attended streams comprise words drawn from either one or two semantic categories, and from categories with either a small or large number of words. Effects of these manipulations on target hit rate and false alarm rate are discussed in relation to previous work.

3aPPb6. A detection theoretical framework for conceptualizing the bottom-up and top-down processes during concurrent-source segregation. Yi Shen (Dept. of Cognit. Sci., Univ. of California, Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697, shen.yi@uci.edu)

During a behavioral task involving the segregation of two concurrent sound sources, bottom-up and top-down processes could both influence the task performance. Therefore, the task performance alone is not assured to provide a complete description of perceptual segregation and does not allow the inference of the contributions from the bottom-up and top-down mechanisms. In the current study, a modeling framework for the perceptual segregation of concurrent harmonic complexes was proposed. The modeling framework allowed the experimenter to probe the status of the segregation, including (1) whether concurrent harmonic complexes were perceived as segregated sources, (2) which one of the segregated sources was used to perform the experimental task, and (3) whether the listener was able to correctly identify the perceived sources. In two experiments, listeners detected changes in the spectral shape of a target harmonic complex when a masker complex was simultaneously presented. By fitting the proposed model to behavioral data, the status of perceptual segregation, usually hidden from the experimenter, could be revealed. Furthermore, the proposed model was able to quantitatively predict the effects of fundamental frequency differences and target-to-masker ratio on concurrent profile analysis using a small number of interpretable parameters.

3aPPb7. Rapid binaural processing for source segregation and lateralization. Darrin K. Reed, Angela Josupeit, and Steven van de Par (Univ. of Oldenburg, Achternstrasse 23, Oldenburg 26122, Germany, darrinreed@hotmail.com)

For realistic listening conditions, interaural cues will fluctuate due to the presence of multiple active sources. If it is assumed that the binaural system is sluggish, then the perceived location of the sound input would be an average of the varying interaural cues. If, however, the binaural system is fast enough to assess the rapidly changing interaural differences, then it could be possible for the binaural system to properly identify the spatial position of a target source. Using a continuous, broadband noise stimulus that contained periodically alternating interaural time differences (ITD) and, notably, no monaural cues, we investigated the binaural system's ability to lateralize brief durations of the target ITD. Results show that listeners can lateralize targets for durations of 3–6 ms indicating that the binaural system allows for a segregation and lateralization of the target and interfering noise streams. Furthermore, results indicate that the binaural system mediates the buildup of a modulated stream. A second experiment investigating whether the salience of the target ITD in the aforementioned stimulus depends on the temporal position of the target within the phase of an amplitude modulated envelope revealed that this was not the case.

3aPPb8. Binaural masking release: An increase in workload capacity. Jennifer Lentz (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, jjlentz@indiana.edu) and James Townsend (Dept. of Psychol. and Brain Sci., Indiana Univ., Bloomington, IN)

The following study applied reaction time analyses of workload capacity to tone-in-noise detection for monaural (NmSm), diotic (tone and noise identical at each ears; NoSo), and dichotic (tone is anti-phase but noise is not; NoSTT) conditions. Reaction times allow comparisons between these conditions at the same signal-to-noise ratios (something which cannot often be done using threshold and percent correct) and can also expose dynamic contributions to the release from masking provided by binaural interactions. Here, we apply a reaction-time based capacity coefficient, which provides an index of workload efficiency. We demonstrate that the release from masking generated by the addition of an identical stimulus to one ear (NmSm vs. NoSo) is unlimited capacity (efficiency ≈1), consistent with an independent parallel-channel model. However, the release from masking generated by the anti-phasic tone (NoSo vs. NoSTT) leads to a significant increase in workload capacity (increased efficiency)—most specifically at lower signal-to-noise ratios. These experimental results provide further evidence that configural processing plays a critical role in binaural masking release, and that these mechanisms operate more strongly when the signal stimulus is difficult to detect.

3aPPb9. Cortical neural correlates of the binaural masking level difference (BMLD). Heather J. Gilbert, Trevor M. Shackleton, Katrin Krumbholz, and Alan R. Palmer (MRC Inst. of Hearing Res., University Park, Nottingham NG7 2RD, United Kingdom, trevor@ihr.mrc.ac.uk)

Single-cell responses to binaural masking level difference (BMLD) stimuli were measured in the primary auditory cortex of Urethane-anesthetised guinea pigs. Firing rate was measured as a function of the presentation level of 500 Hz S0 and Sππ pure tone signals in the presence of N0 and Nππ maskers. The maskers were white noise, low-pass filtered at 5 kHz, with a spectrum level of 23 dB SPL. Responses were similar to those previously reported in the inferior colliculus (IC). At the lowest tone signal levels, the response was dominated by the noise masker, at higher signal levels the firing rate either increased or decreased. Signal detection theory was used to determine detection threshold. Very few neurones yielded measurable detection thresholds for all four stimulus conditions, and there was a wide range in thresholds. However, across the entire population, the lowest thresholds were consistent with human psychophysical BMLDs. Tone and noise delay functions could be used to predict the shape of the firing-rate vs. signal-level function. In summary, like in the IC, the responses were consistent with a cross-correlation model of BMLD with detection facilitated by either a decrease or increase in firing rate.

3aPPb10. Binaural masking level difference on hearing impairments compensating by hearing aids. Cheng-Yu Ho (Dept. of Biomedical Eng., School of Biomedical Sci. and Eng., National Yang-Ming Univ., No.155, Sec. 2, Linong St., Beitou Dist., Taipei City 11221, Taiwan, swellfishyu@gmail.com), Shuenn-Tsong Young (Holistic Education Ctr., Mackay Medical College, New Taipei City, Taiwan), Wen-Ying Yeh, and Zhi-Hong Wang (Dept. of Otolarngology-Head and Neck Surgery, Tri-Service General Hospital, Taipei City, Taiwan)

Many researchers reported that the Binaural masking level differences are reduced in pathological conditions, such as sensorineural hearing loss, retro-cochlear hearing loss, central nervous system disease, and so on. In former studies revealed that the amount of BMLDs are about 2–3 dB or in minus. However, for sensorineural hearing loss, the adaptive method may be limited by the hearing thresholds of subjects. Therefore, to eliminate limitation of hearing thresholds, we delivered stimuli by a pair of sound chambers and receiving by hearing aids fitting with NAL-NL1 formula through headphones to subjects. There were eight mild to moderate sensorineural hearing loss subjects (6 females and 2 males, average 62.2 y/o, std. 9.7) participating in this study. With white noise as masker, we conducted pure tone detection thresholds for 125, 250, 500, 1000 and 2000 Hz in S0N0 and SππN0 conditions. The pilot results showed that the threshold differences (S0N0—SππN0) are 4, 6.3, 4.1, 6.9, and 6.3 dB from 125 to 2k Hz. The pilot results revealed that there are BMLDs on SNHLs with compensating by hearing aids. That is to say, this testing method may be used to rule out supra-threshold deficits from sensorineural hearing losses.

3aPPb11. Extent of lateralization caused by interaural time differences of high-frequency click-trains. Regina M. Baumgaertel and Mathias Dietz (Dept. of Medical Phys. and Acoust. and Cluster of Excellence 'Hearing4All', Oldenburg Univ., Carl-von-Ossietzky Str. 9-11, Oldenburg 26129, Germany, regina.baumgaertel@uni-oldenburg.de)

Hafter and Dye [J. Acoust. Soc. Am. **73**, 644 (1983)] measured threshold interaural time differences (ITDs) for high-frequency click-trains. They found that, given a fixed duration, threshold ITDs were better (lower) for lower pulse rates. Studies by other authors measured lateralization for broad band click-trains and were generally not focused on how the low-frequency fine-structure ITD may influence the results. Such an influence may become prominent at ITDs $> 600\ \mu$s when fine-structure ITDs alone are subject to a cue reversal. The current study therefore focused on high-frequency click-trains and a broad range of ITDs (up to at least 2 ms). The extent of lateralization elicited by these click-trains was determined using an acoustic pointer procedure. Interclick intervals of 5, 10, 20, and 50 ms were investigated. Subjects show an increase in lateralization with increasing ITD even when exceeding the physiologically plausible range of 600 μs. Interclick intervals of 5 ms generally cause the lowest extent of lateralization, in line with the threshold ITD data. The results will be discussed in the light of spatial cue enhancement for cochlear implants, assuming a similar extent of lateralization for interaural pulse time differences at low pulse rates.

3aPPb12. Mapping spatial release from informational masking with one or two masker talkers. Eric R. Thompson (Ball Aerosp. & Technologies Corp., 2610 7th St., Bldg. 441, Wright Patterson, OH 45433, eric.thompson.ctr@wpafb.af.mil), Nandini Iyer, Griffin D. Romigh, and Brian D. Simpson (Air Force Res. Labs, Wright-Patterson AFB, OH)

Spatially separating a target talker from a colocated masker can improve intelligibility, an effect referred to as spatial release from masking (SRM). In this study, listeners' ability to identify speech was measured using the Coordinate Response Measure (CRM) corpus with one or two same-sex CRM maskers as a function of the position of the maskers and the target-to-masker ratio. The target and maskers were each filtered into non-overlapping bands to reduce energetic masking. The target was always straight ahead of the listener, and the masker(s) were presented on the horizontal plane at positions from -90° to $+90^\circ$ in azimuth using individualized head-related transfer functions (HRTF). Consistent with prior results, performance is worst when the target and masker(s) are colocated and improves as the distance between target and maskers increases. Also, performance is better when the two maskers are colocated with each other than when they are separated, and is better when the two maskers are both on the same side of the target than when one masker is on each side of the target. The data suggest spatial filtering strategies that listeners may adopt to improve performance in multitalker scenarios.

3aPPb13. The role of amplitude modulation in auditory 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) and Paul W. Anderson (Dept. of Psychol. and Brain Sci., Univ. of Louisville, Louisville, KY)

The ratio of direct to reverberant sound energy (D/R) has been shown to be a primary acoustic cue to perceived sound source distance. Because it is unclear exactly how D/R might be encoded in the auditory system, a variety of more physiologically plausible correlates to D/R have been identified, including: spectral variance, interaural correlation, and temporal cues. Here, following recent neural work by Kuwada and Kim [ARO (2014)], we describe a new correlate to D/R and perceived distance related to the amplitude modulation (AM) depth of the signal at the listener's location. This cue is caused by the change in the modulation transfer characteristics of the room as a function of source distance. Results from an apparent distance estimation task confirm the efficacy of this AM depth cue in a reverberant soundfield (approximate broadband T_{60} = 3 s), when level cues are made ineffective. Distance estimates were found to be more accurate when the source signal (1-octave band of noise centered at 4 kHz) had AM (32 Hz, 100% depth), and this facilitation was only observed in reverberation. The facilitation was most evident for monaural input, indicating that the AM depth cue is likely processed monaurally.

3aPPb14. Spatial release from masking in musicians and non-musicians. Jayaganesh Swaminathan, Christine R. Mason, Timothy M. Streeter, Gerald Kidd, Jr. (Boston Univ., 635 Commonwealth Ave., Rm. 320, Boston, MA 02215, jswamy@bu.edu), and Aniruddh D. Patel (Tufts Univ., Cambridge, MA)

Recent research suggests that musically trained individuals have enhanced speech-in-noise perception, raising questions about the mechanisms underlying these effects. However, to probe the robustness of this finding and to evaluate theories about the possible mechanisms responsible for this performance advantage, it is desirable to examine speech-in-noise perception using a variety of methods. This study assessed the differences in spatial release from masking (SRM) in musicians and non-musicians. Spatially separating a speech target from interfering masker(s) generally improves target intelligibility; an effect known as spatial release from masking. A speech target was presented simultaneously with two or four speech maskers that were either colocated with the target (0° azimuth) or were symmetrically separated from the target in azimuth (±15° for two maskers; ±15° and ±30° for four maskers). Preliminary results for the two-masker condition indicated greater SRM in musicians than in non-musicians with the differences largely driven by lower target-to-masker ratios for musicians in the spatially separated condition. For the four-masker condition the SRMs observed for the musicians and non-musicians were more similar. However, large individual differences in performance were noted particularly for the non-musically trained group. Future research directions will be discussed, to explore the mechanisms behind these effects. Work supported by NIH-NIDCD and AFOSR.]

3aPPb15. Significance of height loudspeaker positioning for perceived immersive sound field reproduction. Antonios Karampourniotis, Sungyoung Kim (Elec., Comput., TeleCommun. Eng. Technol., College of Appl. Sci. and Technol., Rochester Inst. of Technol., 347 culver Rd., Apt. 5, Rochester, NY 14607, tonykaramp@gmail.com), Doyuen Ko (Audio Eng. Technol., Mike Curb College of Entertainment & Music Business, Belmont Univ., Nashville, TN), Richard King, and Brett Leonard (Sound Recording, Schulich School of Music, McGill Univ., Montreal, QC, Canada)

Recently, new multichannel audio formats incorporating height loudspeakers have caught researchers' attention due to their ability to reproduce an immersive sound field. This study investigated the influence of the height loudspeaker positions and their signals on individually perceived sound quality. The authors generated nine-channel sound sources by convolving two anechoic musical pieces with nine selected room impulse responses measured in different distances and heights. In the listening test two layers of loudspeakers were used: the horizontal layer, following the standard ITU-R BS 775 five-channel loudspeaker configuration and the height layer (with elevation of 30°) with a total of twelve loudspeakers, located at ±30, ±50, ±70, ±90, ±110, and ±130 degrees. Four height signals were reproduced through eight different configurations of four height loudspeakers. Twelve listeners participated in the experiment, wherein they were asked to compare the randomly presented eight configurations and rank them based on their individually perceived sound quality. The experimental results indicate that despite the perceptual differences related to the room impulse responses, the perceived overall quality is significantly influenced by the positioning of the four height loudspeakers.

3aPPb16. Availability of envelope interaural time-difference cues does not improve front/back localization of narrow-band high-frequency targets via head movement. Ewan A. Macpherson (Natl. Ctr. for Audiol., Western Univ., 1201 Western Rd., Elborn College 2262, London, ON N6G 1H1, Canada, ewan.macpherson@nca.uwo.ca)

Information about the front/rear location of a sound source is available in the relationship between the direction of head rotation and the direction of changes in interaural time and level differences (ITD and ILD). Our previous results show that such dynamic cues are highly effective for low-frequency stimuli, but minimally effective for narrowband high-frequency stimuli, in which, respectively, ITD and ILD cues are primarily available. In this study, we assessed the possible benefit for dynamic localization of providing more robust envelope ITD cues in high-frequency stimuli. Listeners judged the front/rear location of anechoic free-field stimuli presented over the central portion of a slow (~0.25 Hz), continual, 90-degree head oscillation. Stimuli were bursts of wideband (0.5–16 kHz), low-frequency (0.5–1 kHz), or high-frequency (6–6.5 kHz) random-phase noise or of raised-sine stimuli with exponent 2, modulation frequency 125 Hz, and bandwidth 6–6.5 kHz. Localization accuracy was high for wideband and lowpass stimuli but poor (and similar) for high-frequency noise and raised-sine stimuli, despite listeners' measured ITD JNDs for raised-sine stimuli being significantly lower than for high-frequency noise. The results suggest that neither veridical dynamic ILD nor ITD cues can overcome the erroneous spectral cue for front/back created by narrowband high-frequency stimuli.

3aPPb17. How high frequency envelopes influence spatial localization in rooms. Salwa Masud, Hari Bharadwaj (Dept. of Biomedical Eng., Boston Univ., 677 Beacon St., Boston, MA 02215, smasud@bu.edu), and Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA)

Perception of sound laterality (left-right angle) is mediated by both interaural time differences (ITD) and interaural level differences (ILD). Previous localization studies in anechoic settings consistently show that low-frequency ITDs dominate perception of source laterality. However, reverberant energy differentially degrades ITDs and ILDs; the effects of room reflections on the perceptual weight given to ITDs and ILDs are not well understood. Here, we tested the hypothesis that high-frequency envelope ITD cues are important for spatial judgments in reverberant rooms by measuring the perceived laterality of high-pass, low-pass and broadband sounds. Results show that when ILD cues and ITD envelope cues are both available, reverberant energy has the smallest effect on localization of high-pass stimuli. When ILD cues are set to zero, localization of high-pass stimuli with strong envelopes (i.e., click trains and speech tokens) is also minimally affected by reverberant energy; however, as envelope modulation is reduced, subjects show increasing localization bias, responding towards the center. Moreover, for stimuli with strong envelopes, subjects with better modulation detection sensitivity are affected less by the addition of reverberant energy. These results suggest that, in contrast to in anechoic space, high-frequency envelope ITD cues influence localization in reverberant settings.

3aPPb18. Weighting of interaural time difference and interaural level difference cues in wide-band stimuli with varying low and high frequency energy balance. Ewan A. Macpherson (The Natl. Ctr. for Audiol., Western Univ., London, ON, Canada) and Tran M. Nguyen (Health and Rehabilitation Sci. Graduate Program, Western Univ. , 205 Oxford St. east, London, ON, Canada, tnguy45@uwo.ca)

Wideband stimuli carry both interaural time and level difference (ITD and ILD) sound location cues. Previously, listener weighting of those cues has only been measured for low-pass, high-pass, and flat spectrum wide-band conditions [Macpherson and Middlebrooks, JASA (2002)]. In this study, we determined how weighting of ITD and ILD cues varied with the low- and high-frequency energy balance in wide-band stimuli. Listeners reported locations of targets that were presented over headphones using individual head related transfer functions. ITD and ILD cues were manipulated by attenuating or delaying the sound at one ear (by up to 300μs or 10 dB), and the final weight was computed by comparing the listener's localization response bias to the imposed cue bias. Stimuli were 100-ms bursts of noise whose spectra were flat from 0.5 to 2 kHz and from 4 to 16 kHz with a level difference between those low- and high-frequency ranges varying in 10-dB steps from −30 to + 30 dB. ILD weight increased (from ~0.5 to 1.5 deg/dB) with increasing high-frequency energy, but ITD weight was constant (~0.08 deg/us) across spectral profiles. The results suggest that in wideband stimuli, weighting of ILD is more stimulus dependent than weighting of ITD.

3aPPb19. The role of interaural level differences in the localization of low-frequency sine tones. Brad Rakerd (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., East Lansing, MI 48824, rakerd@msu.edu), Zane D. Crawford, and William M. Hartmann (Phys. and Astronomy, Michigan State Univ., East lansing, MI)

Five human listeners reported the azimuthal locations of low-frequency sine tones presented in free field, either by a 180-degree loudspeaker array or by virtual reality. The virtual sources were synthesized using cross-talk cancellation based on signals continuously monitored in the listener's ear canals. The experiment tested the duplex model of sound localization, especially the role of interaural level differences (fixed ILD = 0, 6, or 12 dB) for frequencies of 750 Hz or less. Trials with real sources, baseline virtual sources, and virtual sources with manipulated ILDs were always combined in all experiments. Experiments showed that the interaural time differences (ITD) dominated most fixed opposing ILDs, as previously reported, only when interaural phase differences (IPD) were less than 90 degrees. When IPDs exceeded 90 degrees, the ITD lost its influence, and localization was dominated by ILDs. Localization judgments never followed IPDs across a 180-degree boundary, except when a zeroed ILD caused judgments to become chaotic. Within the 90-degree IPD limit, judgments for fixed ILD appeared to follow the ITD better than the IPD. Abnormal interaural conditions (e.g., ITDs and ILDs of opposite sign) led to a notable increase in front-back confusions. [Work supported by AFOSR grant FA9550-11-1-0101.]

3aPPb20. Binaural interference with dynamic interaural cues. Jacqueline M. Bibee (Speech and Hearing Sci., Univ. of Washington, Seattle, WA) and G. Christopher Stecker (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu)

Sensitivity to interaural time and level difference (ITD and ILD) in high-frequency amplitude-modulated sound is reduced by simultaneous presented low-frequency "interferers." Often, the magnitude of this "binaural interference" is limited by the salience of the target cue. This study addressed the hypothesis of reduced interference for salient cues by restricting target cues to sound onset (high salience; condition "R0") or sound offset (low salience for ITD, high salience for ILD; condition "0R"). In control condition "RR," cues remained constant over duration. Targets were trains of 16 Gabor clicks (4 kHz carrier frequency, 2 ms interclick interval). In baseline conditions, targets were presented in isolation; in interference conditions, targets were gated simultaneously with a diotic 500 Hz tone. ITD or ILD detection thresholds were measured adaptively. Results demonstrated significant binaural interference across conditions, despite the use of pulsatile modulators and regardless of the presence or absence of onset cues. Ceiling effects resulted in unmeasurably high ITD (but not ILD) thresholds in some conditions. Consistent with the low salience of envelope ITD near sound offset, this occurred for a majority of subjects when interferers were present but onset cues were not. Work supported by R01-DC011548.]

3aPPb21. Temporal weighting of interaural time differences in low frequency noise presented at low signal-to-noise ratio. Anna C. Diedesch and G. Christopher Stecker (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, anna.c.diedesch@vanderbilt.edu)

Discrimination of interaural time difference (ITD) improves with increasing duration of a target stimulus, but more slowly than expected if ITD sensitivity was temporally uniform over the sound duration. Houtgast and Plomp [JASA **44**, 807–812 (1968)] thus argued for nonuniformal temporal weighting of ITD, in which sound onsets dominate listeners' ITD judgments. That theory is well supported by recent work. Additional data reported by Houtgast and Plomp suggest more uniform weighting in the presence of masking noise at 5 dB signal-to-noise ratio (SNR). The current study measured ITD thresholds for 500 Hz octave-band noise targets with ITD fixed over duration (condition "RR") or changing linearly from zero to peak value (condition "0R") or vice versa ("R0"). Targets were presented in the presence or absence of a continuous 500Hz octave-band masker (5dB SNR). Comparison data were obtained using 500 Hz pure-tone targets across identical ITD and masker configurations (Diedesch and Stecker, 2014, Assoc. Res. Otolaryngol.). ITD detection with pure-tone targets did not appear to benefit (in terms of threshold-vs-duration slopes) from masking noise as in Houtgast and Plomp (1968). Other stimulus conditions that more closely replicate the conditions of that study (i.e., noise targets) are discussed. [Work supported by NIH R01 DC011548.]

3aPPb22. The effect of masker spatial uncertainty on sound localization. Brian Simpson (Air Force Res. Lab., 2610 Seventh St., Area B, Bldg. 441, Wright-Patterson AFB, OH 45433, brian.simpson@wpafb.af.mil), Robert Gilkey (Wright State Univ., Dayton, OH), Nandini Iyer (Air Force Res. Lab., Wright-Patterson AFB, OH), Eric Thompson (Ball Aerosp. & Technologies Corp, Fairborn, OH), Griffin Romigh (Air Force Res. Lab., Wright-Patterson AFB, OH), and Douglas Brungart (Walter Reed National Military Medical Ctr., Bethesda, MD)

Previous research from our laboratory has shown that uncertainty about the spatial location of a masking sound (randomly selected from 1 of 239 locations) can dramatically reduce localization accuracy for a simultaneous target relative to the case in which the masker location is known exactly. One possibility is that knowing the masker location enables the listener to establish a spatial attention filter at that location to suppress the masker and better localize the target. In this experiment, the level of masker spatial uncertainty was systematically varied across blocks by varying the number of possible masker locations (1, 2, 4, 8, or 239) and informing subjects about these possible locations prior to the start of each block. Localization errors were found to increase systematically in the left/right, front/back, and up/down dimensions as the number of potential masker locations increased; this effect was most prominent in the left/right dimension, where localization errors increased by nearly 30 degrees across conditions. Moreover, for a masker in a given location, errors generally increased across these levels of masker spatial uncertainty, consistent with the notion that there is a cost to distributing attention across multiple locations.

3aPPb23. Just noticeable difference of source-receiver distances in the auralization process using speech and music signals. Bernardo Murta, Priscila Wunderlich, Jessica J. Lins de Souza, and Stephan Paul (Undergrad. Program in Acoust. Eng., Federal Univ. of Santa Maria, Av. Roraima 1000, Santa Maria, RS 97105900, Brazil, bernardo.murta@eac.ufsm.br)

The precision of source-receiver transfer functions is of importance to provide reliable and ecologically valid results in auralization. Recently the overall just noticeable difference (jnd) in signals obtained from the convolution of music with slightly varying source-receiver transfer functions was determined using a paired comparison procedure (sound are equal/are different) and estimating the jnd from the 75% point on the psychometric curve. JNDs were 3.55 cm for dislocations approaching the source and 3.46 cm if going away from the source. Now we present the results from tests using a speech signal convolved with the same simulated impulsive responses of different source-receiver positions making 3 sets of 23 stimuli pairs. Each one represents receiver dislocations in one dimension: x, y, and z. The jnd using a speech signal was found to be 3.86 cm when approaching the source and 3.97 cm for the opposite direction. The difference calculated between the jnd of the two tests was 0.314 cm (approaching the source) and 0.506 cm (in the opposite direction). Further tests will be done using noise signals and an overall jnd will be computed.

3aPPb24. The lowest signal-to-noise ratio at which the precedence effect operates for signals in noise. Fiona Guy (MRC/CSO Inst. of Hearing Res. - Scottish Section, New Lister Bldg., Glasgow Royal Infirmary, Glasgow, United Kingdom, fiona@ihr.gla.ac.uk) and Michael Akeroyd (MRC/CSO Inst. of Hearing Res. - Scottish Section, Glasgow, Strathclyde, United Kingdom)

The precedence effect is a robust auditory phenomenon which aids localization of sound in a reverberant room. Its primary characteristic is that the perceived location of the sound is based on the first arriving, direct, sound, with subsequent reflections mostly ignored. Most research into the precedence effect uses stimuli presented in quiet. Here, we used signals in noise, and considered what is the lowest signal-to-noise ratio (SNR) at which the precedence effect remains operative. In our experiment, the stimuli were speech or click trains, presented as direct + reflection pairs over headphones. Crucially, the ITDs of the stimuli set were such that, when presented alone, the direct signal was lateralized to one side of the head but the reflection to the other. A two-interval, two-alternative, lateralization task was used to measure performance. If the reflection was truly ignored, then the perceived lateralization of the direct + reflection pair would give the correct answer to a trial, but if the reflection was not ignored, then the perception would give the wrong answer. The task is SNR dependent, allowing a "precedence threshold" to be measured using an adaptive procedure that varied the SNR of the direct + reflection pair in noise. We report results from both normal and hearing-impaired listeners. [Work supported by the Medical Research Council and the Chief Scientist Office, Scotland.]

3aPPb25. Temporally diffusive reflections and the precedence effect. M. Torben Pastore (Architectural Acoust. and Ctr. for Cognition, Commun. and Culture, Rensselaer Polytechnic Inst., 4 Irving Pl., Troy, NY 12180, m. torben.pastore@gmail.com), Jens Blauert (Inst. of Commun. Acoust., Ruhr-Universitãt , Bochum, Germany), and Jonas Braasch (Architectural Acoust. and Ctr. for Cognition, Commun. and Culture, Rensselaer Polytechnic Inst., Troy, NY)

In reverberant conditions, humans routinely demonstrate an ability to form the auditory event in the direction of the first wavefront and thus identify the direction of the (physical) sound source—the so-called "Precedence Effect." Within limits, this effect even holds when the reflected sounds (the lag) contain more energy than the direct sound (the lead). Previously, we investigated the lateral extent of the Precedence Effect for specular reflections. The current research extends this inquiry by investigating the effect of temporally diffusive reflections, using the same stimuli, but convolving the lag with a 2-ms Hanning windowed Gaussian noise. The lead and lag stimuli were 200-ms Gaussian noise (500-Hz center frequency, 800-Hz bandwidth) presented dichotically with a programmable amount of temporal overlap. The lag/lead level ratio was increased in 2-dB steps, the lead/lag interval was varied from −5 to 5-ms in steps of 1-ms. Listeners indicated the lateralization of their auditory events with an acoustic pointer. The resulting temporal smearing substantially decorrelates lead and lag while keeping them essentially related. The Precedence Effect is found to be more robust to increases of the lag level for diffusive stimuli than for the previously tested sets of specular reflections.

WEDNESDAY MORNING, 7 MAY 2014 — BALLROOM B, 11:00 A.M. TO 12:00 NOON

Session 3aPPc

Psychological and Physiological Acoustics: Auditory Neuroscience Prize Lecture

Alfred L. Nuttall, Chair

Dept. Otolaryngology, Oregon Health & Sci. Univ.,, 3181 S.W. Sam Jackson Park Rd., Portland, OR 97239

Chair's Introduction—11:00

Invited Paper

11:05

3aPPc1. The role of physics in inner-ear physiology and auditory perception. Egbert de Boer (Audiol., Academic Medical Ctr., Meibergdreef 9, Amsterdam 1105AZ, Netherlands, e.d.boer@hccnet.nl)

Auditory analysis of acoustical stimuli has mainly been connected with Fourier analysis. This touches the basic link between auditory perception and the operation of the hearing organs, namely, physics and mathematics. This relation has amply been demonstrated by the work and ideas of Georg von Békésy. Later, the healthy cochlea (inner ear) was found to contain amplifying elements that boost frequency selectivity. Associated with this there is a pronounced nonlinearity. Furthermore, the ear does not only absorb and process sounds, it also emits sound waves. Mathematical models of all these processes must contain subsets serving them all, a goal that has not yet been reached. On the contrary, recent findings on the movements of the various membranes and cells in the organ of Corti have increased the difficulties of mathematical modeling. Additional subjects to be covered in the lecture are (1) in psychophysics of hearing: pitch perception, inharmonic sounds, critical bands, hearing of patients with hearing loss, and auditory revalidation, and (2) in auditory neuroscience: recording of single fibers of the auditory nerve, reverse correlation, inverse solutions of cochlear mechanics, nonlinear analysis, and optical coherence tomography (OCT).

Session 3aSAa

Structural Acoustics and Vibration, Underwater Acoustics, and Physical Acoustics: Session in Honor of Murray Strasberg

David Feit, Cochair
ASA, 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502

Dean Capone, Cochair
Penn State, PO Box 30, State College, PA

Chair's Introduction—8:25

Invited Papers

8:30

3aSAa1. Murray Strasberg, a role model. David T. Blackstock (Appl. Res. Labs. & Dept. of Mech. Eng., Univ. of Texas at Austin, PO Box 8029, Austin, TX 78713-8029, dtb@austin.utexas.edu)

Murray Strasberg held several different offices in the Acoustical Society: Executive Council Member 1969–1972, President 1974–1975, and Secretary 1987–1990. The Society was very fortunate to have Murray at its helm in 1974–1975. Wallace Waterfall died early in Murray's term. ASA old timers remember Wallace as the man who "ran the Acoustical Society out of his hip pocket" for years and years. How were we going to get along without him? Murray led us painstakingly through the difficult reorganization. Also up was the Standards issue. ASA decided to write and publish its own acoustical standards rather than leave them to other organizations. Finally, the recent birth of INCE had sadly led to turf wars, which had to be dealt with. Several years later, long after most presidents have happily gone out to pasture, Murray again stepped up, in this case to fill the void caused by Betty Goodfriend's retirement as ASA Secretary. Murray served as ASA's last Secretary. Beyond his splendid service to the Acoustical Society, Murray was a renowned acoustical scientist and, perhaps above all, a warm and generous friend.

8:50

3aSAa2. Murray Strasberg—A lifetime dedicated to the Acoustical Society of America. Charles E. Schmid (Acoust. Society of America, 365 Ericksen Ave., Bainbridge Island, Washington 98110, ceschmid@att.net) and Elaine Moran (Acoust. Society of America, Melville, NY)

Murray Strasberg received the Gold Medal in 2000 "for contributions to hydroacoustics, acoustic cavitation and cavitation noise, and for dedicated service to the Society" when he was 83 years old. He should have received it much earlier, but the classified nature of his work prevented the preparation of a complete nomination dossier. In addition, most members did not fully realize how much Murray did for the Society as his contributions were spread over a half-century. While serving as President (1974–1975), the Society was faced a critical emergency when Wallace Waterfall, the Society's mainstay and current treasurer passed away. Murray stepped in and worked with others to handle the financial operations of the Society until a new Treasurer was selected. He also served in many other Society roles including Associate Editor of JASA, chair of the committee that set the groundwork for the Congressional Science and Engineering Fellowship Program, and as Secretary from 1987 to 1990. We will share some personal stories about Murray recalled from the many decades working with him as a colleague and wise counsel.

9:10

3aSAa3. Murray Strasberg's contributions to acoustic cavitation and bubble dynamics. Lawrence." Crum (Appl. Phys. Lab., CIMU, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, lac@apl.washington.edu)

Although Murray Strasberg's most significant contributions to acoustics were classified and not available to the general public, he also published some significant papers in the general area of acoustic cavitation and bubble dynamics. Some of these papers are so fundamental in scope that I consider them required reading for any new graduate student that joins our group. At cavitation sessions in the ASA, Murray was always asking insightful questions that amazed others who did not know of his pioneering work on this topic. In this presentation, I will describe some of Murray's contributions to bubble dynamics and cavitation as well as relate some personal observations of my interactions with him over our respective professional careers.

9:30

3aSAa4. Murray Strasberg's contributions to the Navy's ship silencing programs. David Feit (ASA, 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502, feit.d@att.net)

Murray's long career with the U.S. Navy, extended over seven decades 1942–2012, and began when he joined the David Taylor Model Basin (DTMB), now the Carderock Division of the Naval Surface Warfare Center), in 1942. Soon after joining DTMB he was assigned to making cavitation noise measurements on World War II submarines or with model propellers in water tunnels. This lead to his long standing interest in cavitation and bubble noise, matters to be discussed by others in this session. I will discuss his work in structural acoustics, which grew out of his involvement with a propeller related signature source and ultimately proved to be one of his most significant contributions to the Navy's submarine silencing efforts.

9:50

3aSAa5. Hydrodynamic noise—Murray Strasberg's legacy. William Blake (Adjunct Prof. Dept. Aero. Mech. Eng., Univ. of Notre Dame, 6905 Hillmead Rd., Bethesda, MD 20817, hydroacoustics@aol.com)

The decade of the 1950s provided us with the beginnings of sub-disciplines of acoustics that we now call aeroacoustics and hydroacoustics. In the beginning, the attention was placed on mechanisms relevant to the aeronautical engineering and the early rocket and space vehicle communities. Accordingly, much published work at the time dealt with jet noise and structural fatigue resulting from that noise and from turbulent boundary layer excitation. In 1956, Murray Strasberg and Hugh Fitzpatrick published a seminal paper, "Hydrodynamic Sources of Sound", 1st Hydrodynamics Symposium. This paper put the aerodynamic noise theory of Lighthill (1952) in the context of Navy application and defined relevant source types. A. Prosperetti will discuss Murray's legacy regarding bubble noise and cavitation discussed in that paper. I will discuss the other interest of Murray: i.e., flow induced vibration and sound. Although he published little on this subject the impact that he had on others who did was important. Accordingly, Murray had continuing impact on the developing knowledgebase of flow-induced sound and vibration, and we will use the area of TBL noise as an example of how concepts in flow noise and vibration have evolved under Murray's career span.

10:10–10:30 Break

10:30

3aSAa6. Murray Strasberg and bubble acoustics. Andrea Prosperetti (Mech. Eng., Johns Hopkins Univ., 223 Latrobe Hall, Baltimore, MD 21218, prosperetti@jhu.edu) and Andrea Prosperetti (Appl. Sci., Univ. of Twente, Enschede, Twente, Netherlands)

Murray Strasberg made seminal contributions to the nucleation and acoustics of bubbles. Half a century after publication, these papers still receive a sizable number of citations every year. The talk will review this work, comment on its impact, and put Strasberg's classical results in a modern perspective.

10:50

3aSAa7. Oscillations of nonspherical and spherical bubbles. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu), Thomas J. Asaki (Mathematics Dept., Washington State Univ., Pullman, WA), and David B. Thiessen (Chemical Eng. and Bioeng. Dept., Washington State Univ., Pullman, WA)

Murray Strasberg maintained an interest in bubble oscillations throughout his career and frequently offered an incisive perspective. His early bubble research includes [Strasberg, J. Acoust. Soc. Am. **25**, 536–537 (1952); Strasberg, Acustica **4**, 450 (1954)]. His pertinent application of an electrostatic potential theory analogy expands interests of some of James Clerk Maxwell's students. In addition some related discussions with Murray will be recalled and experiments from the 1980s and 1990s at Washington State University pertaining to bubbles and their associated dynamics reviewed. Those experiments concern the response of bubbles to steady and modulated optical radiation forces [Unger and Marston, J. Acoust. Soc. Am. **83**, 970–975 (1988); Unger and Marston, Ocean Optics IX, Proc. SPIE **925**, 326–333 (1988)] and to modulated acoustical radiation forces [Asaki *et al*., Phys. Rev. Lett. **75**, 2686–2689, (E) 4336 (1995); Asaki and Marston, J. Fluid Mech. **300**, 149–167 (1995); Asaki and Marston, J. Acoust. Soc. Am. **102**, 3372–3377 (1997)]. The latter papers pertain to the simultaneous measurement of the frequency and damping of bubble shape oscillations. [Work supported by ONR.]

11:10

3aSAa8. Low-frequency propeller forces and sound—Murray Strasberg's legacy. Jason Anderson (NSWC Carderock Div., 9500 MacArthur Blvd., West Bethesda, MD 20817, jason.m.anderson@navy.mil)

In the early 1960s, Murray Strasberg, after having already made significant contributions to the field of hydroacoustics in the areas of propeller cavitation noise and general hydrodynamic sources of sound, turned his attention toward low-frequency forces that are generated by propellers, which ingest non-uniform flow. Dr. Strasberg developed an elegantly simplistic approach to estimate the propeller unsteady thrust by actually measuring the sound pressure radiation from a prototype propeller installed in a wind tunnel test section. Murray's approach required that the propeller be acoustically compact, thereby allowing the propeller to be treated as a single dipole sound source. This approach offered by Murray, for which a U.S. patent was awarded in 1982, served as a simpler alternative to the more traditional and complicated approach of force dynamometry. Dr. Strasberg focused his efforts on measuring propeller thrust at the blade passage frequency tonals, but he did, however, encourage Maurice Sevik to perform his seminal propeller turbulence ingestion thrust research in the 1960s and 1970s, which truly ushered in the modern era of propeller turbulence ingestion research that continues today. In this presentation, we will describe Murray Strasberg's novel propeller unsteady thrust measurement approach, and we will discuss how researchers over the years have built upon Murray's foundational concept.

Contributed Papers

11:30

3aSAa9. Mass sensing using the time domain response of a functionalized microresonator array. Aldo A. Glean, Joseph F. Vignola, John A. Judge (Mech. Eng., Cathilic Univ. of America, 620 Michigan Ave., N.E, Washington, DC 20064, 10glean@cardinalmail.cua.edu), and Teresa J. Ryan (Mech. Eng., East Carolina Univ., Greenville, NC)

Energy flow between size scales in vibrating structures can be used for mass detection. This work considers structures composed of a larger scale primary resonator (transduction element) and a set of substantially smaller attached resonators (sensing elements). Functionalization of the sensing elements in the downscale realm allows for detection of specific chemical vapors, biological agents, etc. Common approaches for this type of mass detection involve monitoring a structure's frequency domain response for downward shifts in resonance frequencies as mass binds to the sensing elements, or inferring mass changes from shifts in response shapes of the sensing elements. Instead of using frequency domain information, this work describes a detection method based on observing time histories of the energy exchange between the transduction and sensing elements. The energy initially introduced to the system at the transduction element is drawn into the sensing elements. Some energy will return upscale at a later time that depends on the system characteristics. This work demonstrates that by functionalizing every second sensing element, the concentration of adhered mass can be related to the profile of energy returned to the transduction element. Sensor limitations and optimized designs based on measurement noise and fabrication tolerances are also reported.

11:45

3aSAa10. Response shaping using a subordinate oscillator array. John A. Sterling (US Navy, 9500 Macarthur Blvd., Bethesda, MD 20817, john.a.sterling1@navy.mil) and Joseph Vignola (Mech. Eng., Catholic Univ. of America, Silver Spring, MD)

Recent research has shown that arrays of small dynamic elements attached to a master structure can be tuned to significantly alter the time or frequency response of the system. Colloquially known as "fuzzy structures," subordinate oscillators have led to applications including damping, radio frequency filtering, energy harvesting, and micro electro-mechanical system (MEMS) chemical vapor sensing. A passive machinery damping system will be designed and tested for silencing properties. The current subordinate oscillator array (SOA) design consists of a plate of sheet metal with arrays of cantilevers machined of similar but different lengths. These cantilevers will have a range of natural frequencies, which correspond to a desired frequency suppression range. When the SOA is mounted to a vibration source, it functions as an acoustic meta-material which traps and dissipates energy. This is accomplished by synchronizing the phase and frequency of the cantilevers with machinery peak amplitude frequencies. By designing cantilevers properly, the SOA acts as a mechanical broadband filter as opposed to a notch filter. The SOA will be tested primarily for vibration suppression performance but also for sensitivity to tolerance and energy storage density.

WEDNESDAY MORNING, 7 MAY 2014 552 A, 8:25 A.M. TO 11:55 A.M.

Session 3aSAb

Structural Acoustics and Vibration and Noise: Environmental Vibration

Calum Sharp, Cochair
Univ. of Salford, The Crescent, Salford M5 4WT, United Kingdom

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

Chair's Introduction—8:25

Invited Papers

8:30

3aSAb1. Draft American National Standard: Methods for measuring the vibration response of the ground. James E. Phillips (Wilson, Ihrig & Assoc., Inc., 6001 Shellmound St., Ste. 400, Emeryville, CA 94608, jphillips@wiai.com)

Working Group 14 of the Acoustical Society of America/American National Standards Institute S2 Mechanical Vibration and Shock Committee (ASA/ANSI S2 WG14) was formed to develop a standard on methods for measuring the vibration response of the ground to be used when assessing potential impact upon vibration sensitive receivers in the vicinity of rail transit systems. The contents of the standard will be based upon industry accepted methods as described in the Federal Transit Administration guidance manual "Transit Noise and Vibration Impact Assessment" (FTA-VA-90-1003-06) while considering new developments since the latest version of the guidance manual was published in May 2006. This paper will describe the background of the methods and the structure, content, and status of the draft standard.

8:50

3aSAb2. Predicting indoor groundborne noise and vibration levels from transit sources. Shannon McKenna (ATS Consulting, 215 N Marengo Ave. Ste. 100, Pasadena, CA 91101, smckenna@atsconsulting.com)

The groundborne noise and vibration impact thresholds in the Federal Transit Administration (FTA) Noise and Vibration Impact Assessment guidance manual apply to indoor spaces. Therefore, the prediction methodology must account for how the building structure affects groundborne vibration through floor resonances and coupling loss and how the building affects groundborne noise through sound absorption or radiation. This presentation will include measurement results that illustrate the variation in building response to vibration. Due to the large number of vibration sensitive receivers that may be adjacent to a proposed transit line, it can be difficult to account for the variation in building response throughout the corridor when only a limited number of measurements sites are available or practical. Case studies will be presented that include recommendations for how measurement results can be incorporated into the prediction procedure.

9:10

3aSAb3. Validation of the Pipe in Pipe vibration software model to determine ground-borne noise and vibration levels above construction tunnels and the determination of end corrections for different train operating scenarios. Graham Parry (Environment, ACCON UK Ltd., Unit B, Fronds Park, Frouds Ln., Aldermaston, Reading, Berkshire RG7 4LH, United Kingdom, graham.parry@accon-uk.com), Steve Summers (Acoust., Anderson Acoust., Brighton, United Kingdom), and David Yates (Acoust., ACCON UK Ltd., Reading, United Kingdom)

The London Crossrail project requires that re-radiated noise from the construction of the tunnels should achieve exceptionally stringent criteria with respect to groundborne noise and vibration. The challenge has been to implement robust vibration modeling for the movement of construction trains within the tunnel in order to derive noise levels within sensitive properties (including recording studios) above the tunnel. The Pipe in Pipe (PiP) software model (which is based on elastic continuum theory) developed by Hunt and Hussein has been modified and refined empirically utilizing the measurement of exceptionally low vibration levels emanating from the temporary underground construction railway track. The paper describes the extensive and challenging monitoring and modeling which been carried out. The study represents the first rigorous method of determining noise and vibration at properties above the tunnel from construction trains for the Crossrail project, with an associated detailed validation exercise in line with ISO 14837-1 'Mechanical vibration—Ground-borne noise and vibration arising from rail systems'. A combination of modeling and vibration measurements were used to inform the predictions of groundborne noise levels, which can be adjusted to replicate the behavior of other track and rail support types.

9:30

3aSAb4. The acceptability of railway induced vibration in residential environments. James Woodcock, Eulalia Peris, Gennaro Sica, Calum Sharp, Andy T. Moorhouse, and David C. Waddington (Acoust. Res. Ctr., Univ. of Salford, Newton Bldg., Salford M5 4WT, United Kingdom, j.s.woodcock@salford.ac.uk)

The aim of the study presented in this paper is to investigate the use of self-reported acceptability for assessing the human response to environmental vibration in residential environments. The human response to environmental stressors such as noise and vibration is often expressed in terms of exposure-response relationships that describe annoyance as a function of the magnitude of the vibration. These relationships are often the basis of noise and vibration policy and the setting of limit values. This paper takes a different approach by expressing exposure-response relationships for vibration in terms of self-reported acceptability. It is argued that exposure-response relationships expressing acceptability as a function of vibration exposure will complement existing relationships for annoyance in future policy decisions regarding environmental vibration. The results presented in this paper are derived from data collected through a large scale (N = 1431) socio-vibration survey conducted in the United Kingdom, the aim of which was to derive exposure-response relationships for vibration in residential environments. The sources of vibration considered are railways and construction.

9:50

3aSAb5. Differences in the human response to freight and passenger railway vibration in residential environments. Calum Sharp, James Woodcock, Eulalia Peris, Andrew Moorhouse, and David Waddington (Acoust. Res. Ctr., Univ. of Salford, The Crescent, Salford M5 4WT, United Kingdom, c.sharp@edu.salford.ac.uk)

The aim of this paper is to quantify and investigate differences in the human response to freight and passenger railway environmental vibration. Data for this research comes from a field study comprising interviews with respondents and measurements of their vibration exposure ($N = 752$). A logistic regression model has been developed to classify measured railway vibration signals in the field study as freight or passenger signals, with a classification accuracy of 96%. Exposure-response relationships for annoyance due to exposure to freight and passenger railway vibration are then determined using an ordinal probit model with fixed thresholds. These exposure response relationships indicate that the annoyance response for exposure to freight railway vibration is significantly higher than that for passenger railway vibration. In terms of a community tolerance level, the population studied is 15 dB (re 10^{-6} m s^{-2}) more tolerant to passenger railway vibration than freight railway vibration. The potential reasons for this difference in the human response are investigated and discussed. Some of the factors that are investigated include time of day effects, sleep disturbance, effects of combined noise and vibration and the effects of social, attitudinal, and demographic factors.

10:10–10:30 Break

10:30

3aSAb6. Vibration, noise, and their interacting contributions toward sleep disturbance. Michael G. Smith, Ilona Croy, Oscar Hammar, and Kerstin Persson Waye (Occupational and Environ. Medicine, The Sahlgrenska Acad. at the Univ. of Gothenburg, Box 414, Gothenburg 40530, Sweden, michael.smith@amm.gu.se)

The market share of goods traffic operating on the European rail networks is expected to almost double from 2001 to 2020. Nocturnal time slots are expected to play an important part in facilitating this increase and as such sleep disturbance in residential areas is expected to be the most significant hindering factor. Little data currently exist that may be utilized to investigate the potential impact. Within the European project CargoVibes we experimentally investigated sleep disturbance due to vibration and noise arising from freight trains. An experimental trial was conducted involving 23 healthy subjects sleeping for six nights; a habituation night, a control night, and four nights with combinations of vibration and noise exposure. The primary objective was to examine the contribution of each exposure, and investigate how vibration and noise contribute individually and simultaneously to human response. The secondary aim was to determine whether increased numbers of events relative to previous work using 20–36 trains further impacted on sleep. Physiological parameters including sleep stage, cardiac activity, and cortical arousals were obtained using polysomnography and questionnaires were administered to obtain sleep quality data. Results indicate that vibration directly contributes to sleep fragmentation. The findings shall be further discussed at the conference.

10:50

3aSAb7. Health based approach for regulations for vibrations from rail traffic. Martin van den Berg (Ministry of Env., Rooseboomstraat 69, Den Haag 2593PB, Netherlands, m.vdb@xs4all.nl)

Several countries adopted regulations for vibrations and a few made this even statutory. Most do however not endeavor to regulate vibrations are make weak references to—old—standards. Partly, this seems due to the perceived complexity, partly to a lack of knowledge, and partly to the feeling that vibrations are somehow less of a problem. In this paper, a procedure is described to derive limit values for rail vibrations according to WHO-rules for the use of epidemiological evidence in environmental risk assessment. Recent developments in the EU-project CARGOVIBES made it possible to get sufficient data to make this possible. Not all elements that are necessary for a stable regulatory system are obtained, but at least politicians may be supported much better in the decisions for a better protection of the population. Worrying gaps in knowledge are the influence of night exposure on health, the interaction with noise exposure and the effectiveness of measures.

Contributed Papers

11:10

3aSAb8. Acoustical characterization of grass-covered ground. Chelsea E. Good, Joseph F. Vignola, Aldo A. Glean, John A. Judge (Mech. Eng., The Catholic Univ. of America, 620 Michigan Ave., NE, Washington, DC 20064, 26GOOD@cardinalmail.cua.edu), Teresa J. Ryan (Eng., East Carolina Univ., Greenville, NC), Jacob Sunny (Mech. Eng., The Catholic Univ. of America, Washington, DC), and Diego Turo (BioEng., George Mason Univ., Fairfax, VA)

An investigation of acoustical properties of soil covered with live grass is presented. Measurements of normal surface impedance of such samples have been performed over a 200–2000 Hz frequency band using a vertical impedance tube. A phenomenological model is used to predict acoustic properties of the samples. In this work, the samples are considered as a three-component system: soil, grass roots, and foliage. Acoustic impedance of this composite material has components resulting from the different constituent elements. In order to differentiate the acoustic absorption contribution of each element, grass that was controlled for both water content and grass blade height was grown. The acoustic contribution of the soil was determined by performing measurements on unseeded ground with an equivalent watering protocol. Contributions of the roots and foliage were determined by making impedance measurements before and after shearing the mature foliage near the soil surface. The effect of water content in the soil was estimated by making measurements of the samples before and after oven desiccation. We show the effects of roots and foliage on acoustic absorption of grass-covered ground and the acoustic parameters of these complex media estimated using an equivalent fluid model.

11:25

3aSAb9. Prediction and assessment of environmental vibrations from railway operations on Marmaray. Mehmet Caliskan and Salih Alan (Dept. of Mech. Eng., Middle East Tech. Univ., Ankara 06800, Turkey, caliskan@metu.edu.tr)

Marmaray project involves upgrading of commuter lines on both sides of Bosphorus with an uninterrupted, modern, high-capacity commuter rail system. The line totals approximately 76 km including an immersed tunnel under Bosphorus. In this study prediction and assessment of environmental vibration levels due to railway traffic along 20-km portion between Gebze and Pendik on the Asian side are presented. Experimentally obtained existing soil-structure coupling and structural amplification factors for the whole line are applied to the theoretically calculated vibration levels for use in assessment studies. Vibration mitigation measures are devised with respect to Turkish Environmental Noise Regulation and criteria by the Federal Transit Administration.

11:40

3aSAb10. Application magnetorheological elastomer to dynamic vibration absorber for vibration reduction by avariable-unbalance excitation. Unchang Jeong, Jin-Su Kim, Jung-Min Yoon, Jae-Eung Oh (Mech. Eng., Hanyang Univ., 222 Wangsimni-ro, Seongdong-gu, Seoul 133-791, South Korea, unchang.jeong@gmail.com)

This paper presents a concept of dynamic vibration absorber (DVA) applied magnetorheological elastomers (MREs) for vibration reduction. Elastic modulus of MRE significantly increases due to the induced magnetic field. Elastic modulus changes the stiffness of DVA. Thus, the DVA can work effectively in a wide frequency range instead of a narrow bandwidth as a conventional dynamic vibration absorber does. Numerical simulations of avariable-unbalance excitation system are used to validate its effectiveness. Thus, the MRE-DVA will be applicable to the vibration reduction.

Session 3aSC

Speech Communication: Topics in Speech Production (Poster Session)

Alexis R. Johns, Chair
Univ. of Connecticut, 406 Babbidge Rd., Unit 1020, Storrs, CT 06269

All posters will be on display from 8:00 a.m. to 12:00 noon, To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10.00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

3aSC1. Temporal characteristics of vowel articulation: Electromyographic investigation of the "articulatory period". Amy LaCross and E. Fiona Bailey (Dept. of Physiol., Univ. of Arizona, 1501 n. Campbell, Rm. 4104, PO Box 245051, Tucson, AZ 85724, lacross@email.arizona.edu)

In speech, articulator movement precedes the acoustic signal [e.g., Meyer (1991); Gracco (1988); Lubker and Gay (1982); Bell-Berti and Harris (1981)]. The onset and offset of movement, i.e., the 'articulatory period' (Bell-Berti and Harris, 1981, p. 13) is said to encompass the acoustic signal however the precise timing and variability of this period are not well understood. Using electromyography (EMG), we examine the onset/offset of the articulatory period as a function of phonetic context. We recorded EMG activity from the posterior and anterior regions of the genioglossus (GG) muscle of the tongue in subjects during the articulation of static vowels, static vowels with initial coronal or palatal fricatives and vowels embedded in the nonce word [ʃpVp]. We show front and high vowels entail earlier onset of GG activation and that vowels preceded by coronal fricatives are associated with significantly earlier muscle activation than vowels preceded by palatal fricatives. We also note timing differences between GG regions. Thus, for these stimuli, posterior GG EMG activation encompasses the duration of anterior GG EMG activation. These findings underscore the dynamic nature of lingual movement wherein regional tongue muscle activities exhibit predictable differences determined in large part by phonetic context.

8:00

3aSC2. On transition between voice registers: Data from high-speed laryngeal videoendoscopy. Gang Chen, Soo-Jin Park (Dept. of Elec. Eng., Univ. of California, Los Angeles, 63-134 Engr IV, Los Angeles, CA 90095-1594, gangchen@ee.ucla.edu), Jody Kreiman (Head & Neck Surgery, Univ. of California, Los Angeles, Los Angeles, CA), and Abeer Alwan (Elec. Eng., Univ. of California, Los Angeles, Los Angeles, CA)

How specific aspects of vocal fold vibration alter voice register has long been a subject of interest. Transitions between vocal registers are often studied using dynamic vocal fold models and electroglottographic signals. Although laryngeal high-speed videoendoscopy has also been applied to study steady-state voice registers, there has been little such investigation on the transitions between registers. In this study, we examined voice register transitions using phonations in which vocal qualities varied continuously in fundamental frequency (F0), loudness, or voice quality (from breathy to pressed). Glottal area measures {open quotient and alternating-current to open quotient ratios [Chen *et al*., JASA **133**, 1656–1666 (2013)]} and acoustic measures (F0, energy, and H1-H2) were studied using simultaneously collected laryngeal high-speed videoendoscopy and audio recordings from 15 subjects. The video recordings were collected transorally at 10 000 frames per second. Glottal area waveforms were extracted using GlotAnTools (Version 5) [Erlangen, Germany], and acoustic measures were gathered using VoiceSauce software and analysis-by-synthesis. Glottal area measures were compared to acoustic measures and their relationships to voice register transitions were explored.

3aSC3. Duration and rise-span on rising pitch in conversational American English: Rises are never both long and large. Joseph Tyler (Dept. of English Lit. and Linguist, Qatar Univ., Doha P.O. Box 2713, Qatar, josephctyler@gmail.com)

Studies on variation in rising pitch in dialects of English have analyzed correlations of rise-start pitch, rise-end pitch, rise-span, rise-onset position (early vs. late) and pitch dynamism for how they correlate with contextual factors like speaker gender, utterance type (question vs. statement), and turn position (turn-medial vs. turn-final) [e.g., Fletcher and Harrington (2001); Ritchart and Arvaniti (2013); Warren (2005)]. Using data from the Santa Barbara Corpus of Spoken American English, this study adds the novel measure of rise duration. Modeling the data with a linear mixed model with random effects for speaker and conversation (using the lmer function in R), neither gender, utterance type, nor turn position predict rise duration. Nevertheless, rise duration does pattern with the size of the rise (rise-span): many rises are both short and small, some are short and large or long and small, but none are long and large. This suggests a phonetic restriction on the distribution of rises, with speakers avoiding the simultaneous extremes of both duration and span. With the same model testing gender, utterance type and turn position as predictors of rise-span, results show women produce larger rises on questions than statements, while men's rises show no difference between questions and statements.

3aSC4. Intrinsic fundamental frequency (F0) in American English vowels is affected by regional variation. Robert A. Fox and Ewa Jacewicz (SPA Labs, Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210-1002, fox.2@osu.edu)

There is a relationship between F0 values and vowel height: High vowels have higher F0 than low vowels. A long-standing debate has centered on whether this intrinsic F0 (IF0) pattern is an automatic consequence of vowel articulation or whether it represents deliberate effort to enhance vowel contrast. We provide new data from regional variation suggesting that IF0 is partly controlled by the speaker. F0 was analyzed in high, mid, and low vowels, stressed and unstressed, in 36 females representing dialects spoken in Ohio, Wisconsin, and North Carolina. A robust finding was that dialects differ in their use of IF0 in stressed vowels: IF0 differences between high and low vowels in North Carolina were significantly smaller than those in Ohio and Wisconsin for measurements at vowel onset, offset, peak F0, and overall F0. High vowels had higher IF0 than low vowels in all dialects, confirming the universal aspect of IF0. However, the magnitude of this difference was dialect-specific, which suggests a relative independence of IF0 from vowel height. The lack of correspondence between dialect-specific formant values and dialect-specific IF0 indicates that the quality of vowels and their IF0 value are two features that need to be learned separately.

3aSC5. The phonetic status of [æ] in Twi: A reanalysis. Charlotte F. Lomotey (Lit. and Lang., Texas A&M University-Commerce, 1818D Hunt st., Commerce, TX 75428, cefolatey@yahoo.com)

In Akan, [æ] is regarded as an allophone of /a/ and described as "a quality that ranges from a front vowel quality close to [ɛ] in the Asante [Twi] dialect, to a more central quality in the Fante sub-dialects in which it occurs" (Dolphyne, 1988, p. 6–7). In fact, the existence of [æ] appears controversial because while it is believed to be an allophone of /a/ in the Twi dialect (Boadi, 1991; Dolphyne, 2006; O'Keefe, 2003); and in some sub-dialects of Fante (Abakah, 1978; Boadi, 1991; Dolphyne, 2006), recent studies show that [æ] is a phoneme in Fante (Abakah, 2002; Lomotey, 2008). The present study extends the results of Lomotey (2008), in which she suggests that [æ] may not be an allophone of /a/ in Twi, but another realization of [ɪ] or [e] (p>0.05). Results of acoustic analysis (duration, F1, F2) of words with [ɪ], [e] and [æ] in Twi showed similarity with Lomotey (2008). Based on the results, I argue that [æ] may be undergoing or has undergone vowel raising or merger (if it ever exists/existed). I also argue that if it exists in Twi, then it might be represented with a different vowel symbol, and not [æ].

3aSC6. Variability in vowel production: Exploring interactions among frequency, neighborhood density, predictability, and mention. Rachel S. Burdin, Rory Turnbull, and Cynthia G. Clopper (Dept. of Linguist, The Ohio State Univ., 1712 Neil Ave., 222 Oxely Hall, Columbus, OH 43210, burdin@ling.osu.edu)

Multiple factors are known to affect vowel production, including word frequency, neighborhood density, close predictability, and mention. In this study, we explored interactions between the effects of all four of these factors on vowel duration and dispersion in a fully crossed within-subjects design. Participants read a series of short stories that contained target words varying in frequency, neighborhood density, predictability, and mention. Vowel duration and dispersion from the center of vowel space were measured. Results from linear mixed effect modeling revealed the expected effect of neighborhood density on duration: vowels in words with higher neighborhood density were longer. Both frequency and mention had significant expected effects on vowel duration and dispersion: vowels in more frequent words and second mention words were shorter and less peripheral. An interaction between frequency and mention on dispersion was also observed, such that low frequency words underwent second mention reduction to a greater degree than high frequency words. No effects of predictability were observed. These results are only partially consistent with reported findings elsewhere in the literature, suggesting that these effects are fragile and may be substantially affected by methodological choices, including the target words, task, and statistical models.

3aSC7. The implementation of voicing in obstruents in American English connected speech. Lisa Davidson (Linguist, New York Univ., 10 Washington Pl., New York, NY 10003, lisa.davidson@nyu.edu)

Descriptions of obstruent voicing in English phonetics textbooks observe that [+ voice] obstruents are likely only fully voiced between other voiced sounds, but have less voicing in word-initial and word-final position [e.g., Cruttenden (2008); Docherty (1992) for experimental data for British English]. This study examines these claims for American English using a corpus of 27 speakers reading 3–5 short stories. Voiced stops and fricatives were coded for preceding and following segments, stress of the previous and following syllables, position within the word (initial, medial, and final), and phrasal position of the word. The calculation of unvoiced frames from the PRAAT voice report was used to obtain the proportion of voicing during each obstruent closure. Results indicate the influence of several phonotactic and prosodic factors. (1) The segment preceding the obstruent influences the amount of voicing more than the following segment does; complete devoicing and shorter durations of partial voicing are most often conditioned by preceding pauses or voiceless sounds. (2) Among following sounds, nasals condition the most complete voicing. (3) An obstruent's position within a word has less effect than its phrasal position. (3) Stress primarily affects stops: a preceding stressed syllable conditions significantly more voicing than a preceding unstressed syllable.

3aSC8. Nuclear pitch accent in American English dialects: Truncation and compression. Ewa Jacewicz and Robert A. Fox (SPA Labs, Speech and Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43210-1002, jacewicz.1@osu.edu)

Dialects of the same language can differ in their use of prosody. We explored realization of nuclear pitch accent in female speech in three regional varieties of American English: Ohio, Wisconsin, and North Carolina. We found that both OH and WI speakers truncate the F0 contours while NC speakers compress them in identical environments. Our second question pertained to the effects of obstruent voicing in syllable coda on the F0 of the preceding nuclear vowel. We admitted three possibilities for the effects of a voiceless coda: shortened vowel duration may (1) "clip" the F0 contour and reduce the dialectal differences; (2) maintain the dialectal differences by preserving the F0 contour shapes found before a voiced coda; or (3) some dialects may "clip" and some may preserve the F0 contour. The results supported the third option. Ohio and Wisconsin speakers "clipped" the F0 contour so that F0 terminal values before a voiceless coda were higher than before a voiced coda. However, North Carolina speakers not only preserved the contours but their F0 terminal values before a voiceless coda were substantially lower than before a voiced coda. The effects of coda voicing on F0 fall were found to augment the dialectal differences.

3aSC9. Ixpantepec Nieves mixtec question intonation. Younah Chung and Amanda Ritchart (Linguist Dept., UCSD, 9500 Gilman Dr., La Jolla, CA 92093-0108, yachung@ucsd.edu)

Ixpantepec Nieves Mixtec (INM) is posited to have a three-way lexical tone system and lexical stress. However, no known work has investigated whether INM makes use of F0 intonationally. Our study addresses this gap and seeks to contribute to the typology of tone-intonation interactions. We recorded both polar and wh-questions. For polar questions, we analyzed the F0 from one female speaker. The speaker produced question-answer (Q&A) pairs, where the object of transitive verb phrases appeared in both the interrogative and declarative sentences, and differed in its lexical tone. The F0 of every vowel was measured to determine if lexical tones vary depending on whether they occur in an interrogative or declarative sentence of the Q&A pair. Pitch raising is the only significant difference ($p < 0.01$) between the Q&A pair; both high and low lexical tones have higher F0 in the interrogative compared to the declarative. For wh-questions, a qualitative analysis was done using interview data from a male and female speakers. A quantitative analysis is also underway. The Q&A pairs had similar pitch contours and absolute F0 values; thus, no wh-question intonation or tone-intonation interaction was found. Implications for intonational typology in languages with lexical tones will be discussed.

3aSC10. Japanese children's suprasegmental imitation capability of English rhythm. Hiromi Kawai (Ctr. for Teaching English to Children, Kanda Univ. of Int.. Studies, 1-4-1 Wakaba, Mihama-ku, Chiba City 2610014, Japan, kawai-h@kanda.kuis.ac.jp)

This study reports on an acoustic analysis of L2 Japanese child learners' English rhythmic imitation capabilities. The participants for this study were Japanese fifth and sixth graders, all of whom had been taking English language classes since their first year of elementary school. Rhythmic chants were employed as a means to elicit oral production owing to their frequent use in the EFL classroom. Nine English sentences (including two to six stressed syllables, four downbeat sentences and five upbeat sentences) were employed in the collection of a rebus rhyme and in consideration of word familiarity. In the first of two parts of the experiment, the participants shadowed back the test sentences following the experimenter's model pronunciation. In the second part, the experimenter showed picture cards related to the test sentences during the participants' imitation of the experimenter's utterance. During both tests, the experimenter tapped rhythm in time to each participant's utterance of stressed syllables instead of using automatic rhythmic machine. Acoustic analyses and Cluster analysis were used to discriminate between the native-like rhythmic and non-native groups, and profile the 5th and 6th grade groups. Overall, the findings have clear implications for the teaching of English rhythm to moraic language speaking children.

3aSC11. An acoustic analysis of advanced tongue root harmony in Karajá. Sean A. Fulop and Ron Warren (Linguist, California State Univ. Fresno, 5245 N Backer Ave., PB92, Fresno, CA 93740-8001, sfulop@csufresno.edu)

Recent phonological analysis of Karajá, a Macro-Jê language of Brazil, claimed that the language's vowel system evinces advanced tongue root (ATR) harmony [Ribeiro, "ATR harmony and palatalization in Karajá," Santa Barbara Pap. Linguist. **10** (2000)]. Despite this, the phonetic facts about these vowels have never been published, which has left the only claim that ATR operates in any American language subject to controversy. The use of tongue root advancement can never be completely established without articulatory investigation (e.g., MRI scan), but we here provide an acoustic analysis of two native Karajá speakers, which examines the correlates of [ATR/RTR] in four pairs of vowels. An ATR vowel involves expansion of the pharyngeal cavity by moving the base of the tongue forward and/or lowering the larynx during vowel production, and being generally involved in vowel harmony, contrasts with a retracted [RTR] version of the vowel. Acoustic correlates of tongue root advancement generally include a lowering of the frequency of F1 as the pharyngeal cavity expands, together with changes in spectral timbre as measured by the relative formant amplitudes. Particularly, the amplitude of F1 is frequently greater in [ATR] vowels when compared with [RTR] vowels [Fulop *et al.*, "An acoustic analysis of the tongue root contrast in Degema vowels," Phonetica **55**, 80–98 (1998)], and this will be used to illuminate the phonological claims.

3aSC12. Phonetic accommodation in Russian-Estonian bilingual speech. Cameron Rule (Dept. of Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan St., Ann Arbor, MI 48109, rule@umich.edu)

The influence of L1 phonological knowledge on L2 bilingual speech has long been a fruitful topic of speech perception and production research (Piske *et al.*, J. Phonetics **29**, 191–215). Recent research examining phonetic accommodation in bilingual speech has demonstrated that L1 and L2 phonologies interact in a bidirectional, dynamic manner (Fowler *et al.*, J. Phonetics **36**, 649–663). My study builds on these results by examining the acoustic outcomes of phonetic accommodation in vowels produced by bilingual Russian(L1)-Estonian(L2) speakers. Specifically, I aim to test the hypotheses that L2 knowledge can significantly affect robust aspects of L1 speech, i.e., vowel quality, and that there is a correlation between the extent of L2 exposure and the degree of phonetic accommodation. The preliminary results of acoustic analysis suggest that bilingual speakers produce L1 vowels with formant values that are intermediate from both L1 and L2 monolingual controls, indicating that L2 exposure influences significant aspects of L1 speech. Presently, I am conducting analyses which compare individual speaker differences in the extent of phonetic convergence to the amount of their L2 use and exposure. My overarching objective is to present novel empirical evidence that expands our current understanding of phonetic accommodation in bilingual speech production.

3aSC13. Intelligibility in dysphonic speech: Landmark-based measures. Suzanne Boyce, Keiko Ishikawa (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, 5371 Farmridge Way, Mason, OH 45040, ishikak@mail.uc.edu), and Marisha Speights (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

Speakers with voice disorders frequently report reduced intelligibility in ordinary communication situations. This effect is typically attributed to reduced loudness; however, other source/vocal tract interactions may be at work. The acoustic landmark theory of speech perception postulates that specific acoustic events, called "landmarks," contain particularly salient information about acoustic cues used by listeners. The current study examined acoustic profiles of dysphonic speech with the publically available landmark-based automatic speech analysis software, SpeechMark^{TM}. In this study, we focused on burst landmarks, which aim to identify onsets and offsets of affricate/stop bursts. The study tested two hypotheses: (1) normal and dysphonic speech samples differ in the number of burst landmarks because laryngeal pathology affects the consistency of airflow control, and (2) the number of burst landmarks will correlate with cepstral peak prominence values, which have been shown to correlate well with perceptual judgment of dysphonia severity. Speech samples of 36 normal and 33 dysphonic speakers from KAY Elemetrics database of Disordered Voice were subjected to the analysis. Results will be discussed in the context of clinical assessment of intelligibility for dysphonic voices.

3aSC14. The merged vowel of PIN and PEN as realized in Bakersfield, California. Ron Warren and Sean A. Fulop (Linguist, California State Univ. Fresno, 5245 N. Backer Ave., Linguist PB92, Fresno, CA 93740-8001, warlockwarren@gmail.com)

One consistently identified quality of Southern American English (SAE) is the PIN/PEN merger, wherein the vowels /ɪ/ and /ɛ/ become indistinguishable in nasalized contexts. Existing studies have shown that the merged vowel may either occupy a space between the vowel spaces of /ɪ/ and /ɛ/ [Bigham, MA thesis, UT Austin (2005)], or merge completely into /ɪ/ [Koops *et al.*, "The effect of perceived speaker age on the perception of PIN and PEN vowels in Houston, Texas," U Penn Working Pap. Linguist. **14**(2) (2008)]. Due to the dustbowl migration, areas of the southern central valley of California have a complete PIN/PEN merger similar to SAE. This study examines the acoustic properties of the merged vowel spoken by eight speakers from Bakersfield, California, ages 25 through 65. Findings indicate that the merged vowel does not occupy a compromise position, nor a merged /ɪ/ position in any but the oldest subject. Instead, the merged vowel presents significantly lower first formant values than /ɪ/, suggesting that as the California vowel shift has lowered the front lax vowels, this merged allophone has remained stationary. Further, the youngest female subjects presented an inconsistent vowel merger, suggesting a potential demerger of the type documented in southern urban centers. This raises questions about the nature of the alleged merger, if perceptible acoustic differences remain which allow for demerger.

3aSC15. Linearly diverging vocal folds and the Coanda effect. Jesse Haas, Xavier Pelorson (Parole et Cognition, INPG, Universite Stendhal, CNRS, Domaine Universitaire, BP46, Saint Martin d'Heres 38402, France, jesse.haas@gipsa-lab.grenoble-inp.fr), and Avraham Hirschberg (Technische Universiteit Eindhoven, Zwartenberg, Netherlands)

We would like to verify whether the Coanda effect has a significant impact when incorporated into theoretical vocal fold models that assume a piecewise linear shape of the vocal fold walls, as many do. We model the intraglottal flow with the equations of Thwaites. Thwaites boundary layer theory gives simple criteria for the glottal jet separation point if the vocal folds diverge linearly, and even validates well-known empirical observations for the jet width at flow separation. We test this criteria against flow experiments with rigid vocal fold replicas. The experiments involve symmetric and asymmetric vocal fold configurations, as well as steady and unsteady flow. We then validate the significance of the predicted Coanda effect on several numerical models of human vocal folds. We test the significance of the effect both on mechanically symmetric vocal fold models and on ones with mechanical asymmetries. We find limited effects on symmetric vocal folds and varying degrees of impact on asymmetric vocal folds.

3aSC16. Realization of stress in theatrical speech: The example of Composed Theater. Kostis Dimos (Phonetic Lab., Dept. of General Linguist, Univ. of Zurich, Plattenstrasse 54, Zurich CH-8032, Switzerland, kostis.dimos@uzh.ch), Leopold Dick (Bern Univ. of the Arts, Bern, Switzerland), and Volker Dellwo (Phonetic Lab., Dept. of General Linguist, Univ. of Zurich, Zurich, Switzerland)

In this study, we investigated syllabic stress in theatrical speech containing emphatically stressed vowels. Typical acoustic correlates of stress are intensity, pitch, and duration. The aim of this experiment was to measure whether artistic emphatic stress is realized differently in terms of these correlates compared to stress in normal speech. In an experiment, one professional performer of Composed Theater, an avant-garde type of experimental theater, was recorded both in natural and theatrical French speech. The results showed an increase in mean intensity, pitch, and mean duration in highly stressed vowels. The difference between stressed and unstressed syllables was higher in theatrical, than in normal speech. We argue that duration, pitch and intensity were used to different degrees to indicate stress in the artistic speech sample. Our study provided a preliminary comparison of speech characteristics between artistic and normal speech focusing on stress realization techniques in artistic speech performance. Additional professional speakers' recordings are currently being analyzed in terms of prosodic characteristics in avant-garde theater performance.

3aSC17. Contextual landmark analysis of speech from typically and atypically developing children. Chelsea Levy, Allison Mann, Jess Kenney, Jeung-Yoon Choi, and Stefanie Shattuck-Hufnagel (Res. Lab. of Electronics, MIT, 50 Vassar St., Rm. 36-581, Cambridge, MA 02139, jyechoi@mit.edu)

Analysis of acoustic landmarks (abrupt changes in the speech signal spectrum, Stevens 2002) has been carried out for speech from two typically and six atypically developing children. The atypically developing children include two with specific language impairment (SLI), and four diagnosed with autism spectral disorder, and two with higher and two with lower language functions (ASDH and ASDL, respectively). Recordings of non-word repetition sessions (CNREP) for each child were hand annotated with words, phones, and landmarks (42 tokens per child). Decision tree analysis was used to examine the effects of factors such as landmark type, preceding and following phone type, position in the syllable (onset, nucleus, ambisyllabic, and coda), and syllabic stress (stressed, full, and reduced) on landmark modification patterns. Results indicate that, compared with typically developing children, atypically developing children show different landmark modifications, and that these different modification patterns can be characterized by the systematic effects of contextual factors.

3aSC18. Measures of spectral tilt in Shanghainese stops and glottal fricatives. Laura L. Koenig (Haskins Labs and Long Island Univ., 300 George St., New Haven, CT 06511, koenig@haskins.yale.edu) and Lu-Feng Shi (Long Island Univ., Brooklyn, NY)

Shanghainese differs from other major Chinese dialects in having a three-way contrast among stop consonants. Although phonological descriptions often label these sounds as voiceless aspirated, voiceless unaspirated, and voiced, in contemporary Shanghainese the "voiced" category is not typically produced with closure voicing in initial position. Rather, the vowel onset appears to be characterized by breathy voicing, as demonstrated by acoustic, aerodynamic, and transillumination data. Past studies have been limited to few speakers. This study presents preliminary data of a large-scale study of the Shanghainese stop system as well as the voicing contrast in glottal fricatives, another unusual feature of the language. Data have been collected from 20 male and female native speakers. Measures are made of voice onset time (for the stops), and three measures of spectral tilt (for stops and glottal fricatives): The relative amplitudes of (a) the first two harmonics (H1-H2); (b) H1 and the first formant (H1-A1); and (c) H1 and the third formant (H1-A3). Spectral tilt measures are taken at phonation onset and 50 ms into the vowel. Preliminary analyses suggest that all three spectral measures may lend insight into the nature of the source distinction among the stops as well as the glottal fricatives.

3aSC19. Assessing the effects of cognitive decline on the speech rate of demented and healthy elderly speakers. Maria Heffinger, Gina DAmico, Linda Carozza (Commun. Sci. and Disord., St. John, Staten Island, NY), Fredericka Bell-Berti, and Pamela C. Krieger (Commun. Sci. and Disord., St. John, 8000 Utopia Parkway, Queens, NY 11439, bellf@stjohns.edu)

According to current research, both age and cognitive decline have an effect on speech rate. While healthy elderly speakers are reported to speak at a slower rate than their younger counterparts, it has also been noted that elderly patients with a cognitive decline speak more slowly than their healthy elderly peers. This study analyzes the speech rates of healthy elderly speakers and compares the results with our previously reported data from speakers diagnosed with dementia. Our healthy elderly participants repeated five-syllable carrier phrases with varying target words. The durations of consonant and vowel segments will be measured using digital spectrograms to determine the extent of final lengthening, compensatory shortening, and overall speech rate (in syllables and second). Results for the two speaker groups will be compared with each other and also with results previously reported in the literature.

3aSC20. An acoustic analysis of lexical stress in non-words in Uyghur. Mahire Yakup (Univ. of Kansas, 2211 Willow Crk, Lawrence, KS 66049, mylyakup@ku.edu)

Previous research examined the stress pattern in Uyghur, a Turkic language, using real words and found that Uyghur used the duration as a stress cue but not F0, and the intensity was moderated by syllable types. In order to avoid vowel lengthening in the real words, the non-words (DOSdos vs. dosDOS or DOdo vs. doDO, capitalized as stressed) were used, in which the syllable types (CVC syllable and CV syllable) and vowels (a, u, i, o, ø, y) were controlled. In the production experiments, speakers were clearly instructed that the capitalized ones were stressed and lower-cased as unstressed syllables. All target words were embedded in the same carrier sentence (e.g. 'I will say X now.'). In the production of ten female native Uyghur speakers, average fundamental frequency, duration, average intensity, and first and second formant frequencies for vowels were collected in the accented and unaccented syllables. The results showed that there were significant differences in duration and intensity between stressed and unstressed syllables. The fundamental frequency differences were associated with word final positions rather than stress positions. Vowels are centralized in the unstressed position. The present acoustic data suggest that native Uyghur speakers use of duration and intensity rather than F0.

3aSC21. Effects of predictability on vowel reduction. James D. Whang (Linguist, New York Univ., 10 Washington Pl, New York, NY 10003, james.whang@nyu.edu)

High vowel devoicing in Japanese, where unaccented /i, u/ in a C_1VC_2 sequence reduce when both C_1 and C_2 are voiceless, has been studied extensively, but whether the target vowel is truly devoiced (oral gesture is maintained) or deleted completely (oral gesture and voicing are both lost) is still debated. This study examines the effects of vowel predictability on the degree of vowel reduction. Native Japanese speakers ($N = 8$) were recorded in a sound-proof booth reading sentences containing lexical stimuli. C_1 of the stimuli were [k, ʃ], after which either high vowel can occur, and [ϕ, s, ç], after which only one of the two is possible. Half of the stimuli contained a devoicing environment with a voiceless C_2. Center of gravity, the amplitude weighted mean of frequencies present in a signal, was measured for the first half (COG1) and the second half (COG2) of C_1. Results show that COG2 is significantly lower than COG1 for all consonants when a full vowel follows, as well as for [k, ʃ] in devoicing stimuli. In contrast, COG remained stable for [ϕ, s, ç] in devoicing environments, suggesting a complete lack of vowel gestures. Predictable vowels, therefore, seem to delete, while unpredictable vowels devoice.

3aSC22. Lung volume initiation levels and selected acoustic measures of English consonants. Peter J. Watson and Yu-Wen Chen (Speech-Language-Hearing Sci., Univ. of Minnesota - Twin Cities, 164 Pillsbury Dr., Shevlin 115, Minneapolis, MN 55455, pjwatson@umn.edu)

Watson, Ciccia, and Weismer (2003) described the relationship of initiating speech at different lung volumes to selected acoustic variables related to vowel production. It was found that some variables, such as dB SPL were related to a 'direct' mechanical interaction with the breathing system. They also found that a variable, such as vowel-space, was reduced at low-lung volume initiation levels suggesting an indirect link between one subsystem, breathing, and another articulation. Using a similar procedure to Watson *et al.* (2003), we studied selected acoustic variables of English fricatives and stops. Participants read aloud a carrier phrase with a 2 syllable V—'CV embedded within it. Participants were trained to initiate speech at 3 different lung volume levels: normal, low, and high. Data will be discussed in relation to those acoustic variables that have a more direct interaction with the breathing system, e.g., db SPL, and those with a less direct relationship with breathing, e.g., relative duration of voice-onset-time between cognate pairs of voiced and voiceless stop consonants, and the difference of first moment measures between sibilants.

3aSC23. Effects of speaking mode (clear, habitual, slow speech) on vowels of individuals with Parkinson's disease. Rebekah A. Buccheri (Dept. of Speech-Lang. Pathology/Audiol., Molloy College, 1000 Hempstead Ave., W104, Rockville Ctr., NY 11571, rbuccheri@molloy.edu), Douglas H. Whalen, Winifred Strange (Speech-Language-Hearing Sci., The Graduate Ctr. (CUNY), New York, NY), Nancy S. McGarr (Dept. of Speech-Lang. Pathology/Audiol., Molloy College, Rockville Ctr., NY), and Lawrence J. Raphael (Commun. Sci. and Disord., Adelphi Univ., Garden City, NY)

This study examined the effects of three different speaking modes (clear, habitual, and slow speech) on speech production of individuals with and without Parkinson's disease. Twenty-one speakers (13 with Parkinson's, 8 Controls) read the Farm passage in habitual, clear, and slow speech modes. Acoustic analysis involving the assessment of the first and second formant frequencies was performed using: vowel space areas, vowel dispersions, /i-ɑ/ distances measures for both tense and lax vowels produced in each of the speaking conditions. Results revealed that for both groups, the vowel space areas were larger in the clear and slow conditions compared to habitual, with no difference between clear and slow for tense vowels. However, there was no significant difference across any of the speaking conditions for lax vowels. There was a significant difference in vowel dispersion measures, between the habitual and clear speech conditions and also the slow and habitual conditions. But there was no significant difference between the clear and slow conditions for vowel dispersions. With respect to /i-ɑ/ distance, there was a significant difference between the habitual condition and clear speech. Implications of these results will be presented for both Parkinson's disease subjects and normal controls.

3aSC24. Segmental and prosodic effects on intervocalic voiceless stop reduction in connected speech. Dominique A. Bouavichith (Dept. of Linguist, New York Univ., 33 Washington Sq W, 1115, New York, NY 10011, dab491@nyu.edu)

Descriptions of English and other languages have claimed that voiced and voiceless intervocalic stops are often lenited to fricatives and approximants in connected speech. Few acoustic analyses of factors that affect this reduction have been reported for American English [cf. Lavoie (2001), Tucker and Warner (2011)]. In this analysis, intervocalic voiceless stops produced in bisyllabic words during story reading are examined (participants N = 19). The first result shows that speakers never lenite voiceless stops to approximants, except when /t/ is produced as the approximant implementation of a flap. This shows that voiced and voiceless stop reduction behave differently. Second, stress and vowel reduction play an integral role: 33% of stops are produced as fricatives when stress is on the preceding syllable (e.g., "taco"); 4% when stress is on the following syllable (e.g., "account"). The rate of reduction is significantly lower when stops are surrounded by two full vowels and higher when they are followed by schwa. Third, fricative reduction is most common for /k/, since full closures may be most difficult in the velar region. This study will be compared to Bouavichith and Davidson's (2013) examination of voiced stops to provide a fuller picture of intervocalic stop reduction in American English.

3aSC25. Temporal alignment between head gesture and prosodic prominence in naturally occurring conversation: An electromagnetic articulometry study. Dolly Goldenberg (Linguist, Yale Univ., 192 Foster St., Apt. 1, New Haven, CT 06511, dolly.goldenberg@yale.edu), Mark Tiede, Douglas N. Honorof (Haskins Labs., New Haven, CT), and Christine Mooshammer (Institut für deutsche Sprache u. Linguistik, Berlin, Germany)

Studies of the relationship between speech events and gesticulation have suggested that the peak of the prosodic pitch accent serves as a target with which body gestures may be coordinated (Roth, 2002; Loehr, 2004). While previous work has relied on controlled speech elicitation generally restricted to nonrepresentational extension/retraction (Leonard and Cummins, 2011) or iconic (Kelly *et al.*, 2008) gestures, here we examine the kinematics of the speech articulators and associated head movements from pairs of individuals engaged in spontaneous conversation. Age and gender matched native speakers of American English seated 2 m apart were recorded using two electromagnetic articulometer (EMA) devices (Tiede and Mooshammer, 2013). Head movements were characterized by the centroid of reference sensors placed on the left and right mastoid processes and the upper incisors. Pitch accents were coded following the ToBI implementation of Pierrehumbert's intonational framework following Beckman and Elam (1997). Preliminary findings show that the apex (point of maximum excursion) of head movements within an IP in general precedes the peak of the associated pitch accent, and is consistently aligned with co-occurring articulatory events within the syllable. [Work supported by NIH NIDCD-DC-012350.]

3aSC26. Production and perception of the English sibilants /s/ and /ʃ/ in persons with Parkinson's disease. Yu-Wen Chen and Peter J. Watson (Speech-Language-Hearing Sci., Univ. of Minnesota, 164 Pillsbury Dr. SE, Shevlin Hall 115, Minneapolis, MN 55455, chen1887@umn.edu)

Parkinson's disease (PD) presents with both sensory and motor deficits including speech, characterized by soft voice, monotonicity, and imprecise articulation. Both audition and somatosense are believed important for accurate speech production. Individual studies have identified auditory impairment and somatosensory impairment of the orofacial structures and the larynx. No studies have examined the relationship between imprecise articulation and sensory impairment. This research aims at examining sensory deficits and the relationship between sensory deficits and imprecise articulation in persons with PD. The production of /s/ and /ʃ/, auditory discrimination and identification in spectral shape, and somatosensory acuity on the tongue tip are examined. In the production task, participants read monosyllabic words beginning with /s/ and /ʃ/ embedded in a carrier sentence; measures of spectral shapes are made. Auditory tasks comprise an acuity task testing for participants' auditory acuity in perceiving spectral shapes and a /s/-/ʃ/ categorization task testing for their categorical boundaries. The somatosensory acuity task tests for participants' somatosensory acuity in spatial orientation on the tongue tip. Relationships between production of /s/ and /ʃ/ and sensory performance are tested in correlational tests.

3aSC27. Relationship between the first two formant frequencies and tongue positional changes in production of /aɪ/. Jimin Lee (Commun. Sci. and Disord., Penn State Univ., 404A Ford Bldg., University Park, PA 16802, jxl91@psu.edu)

The first two formant frequencies (F1, F2) of vowels are often interpreted in terms of their relationship to tongue height and advancement, respectively. To test this interpretation, the current study examines the relationship between F1/F2 trajectories and tongue positional changes in production of the diphthong /aɪ/ by utilizing electromagnetic articulography. Ten healthy female speakers participated in the current study. Electromagnetic articulography (AG-200) and a synchronized audio recording system were utilized to obtain synchronized kinematic and acoustic data. Each speaker produced three repetitions of the word "hide" in the carrier phrase "I say a__ again." F1 and F2 were traced along the entire vocalic nucleus of the target vowel /aɪ/. Over the same time interval, x and y coordinate values from the tongue sensor (positioned approximately 25 mm away from tongue apex) was recorded. Correlational analysis (r-values) showed that, overall, F1—tongue y position and F2—tongue x position pairs have strong relationships; however, a strong relationship was observed in F1-x and F2-y pairs as well. Results will be discussed in terms of amount of variance of formant frequencies explained by tongue xy positional changes and issues of interpretation of formant frequencies.

3aSC28. Vowel and consonant effects on subglottal pressure. Didier Demolin, Silvain Gerbers (Gipsa-lab, Université Stendhal, 1180 Ave. Centrale, Grenoble Grenoble cedex9, France, didier.demolin@gipsa-lab.grenoble-inp.fr), and Sergio Hassid (Gipsa-lab, Université Stendhal, Brussels, Belgium)

The respiratory system is generally regarded as producing voluntary variations in intensity and perhaps in pitch, but not producing voluntary increases in pressure for particular sounds. All the changes related to individual segments, such as the drop in subglottal pressure that occurs after [h] or the increase in pressure during the [k] closure are considered to be aspects of tract aerodynamics, and not under voluntary control. They can be

ascribed to variations in the resistance provided by the vocal folds to the outgoing air (the glottal impedance) or to variations in the stiffness of the vocal tract walls. This paper examines variations accompanying different vowels and consonants, and concludes that it is not only the Koreans who use greater respiratory effort to distinguish some sounds. The principal finding of this study was that there was a considerable difference in the subglottal pressure associated with the different vowels. Across all other variables, the difference in subglottal pressure between /a/ and /i/ is 1.65 cm hPa and between /a/ and /u/ is 1.76 hPa. Both these differences are highly significant ($p < 0.0001$) in a one way ANOVA. The difference between /i/ and /u/ is 0.11 hPa and is not significant ($p = 0.303$).

3aSC29. Phonation and tone in conversational Beijing Mandarin. Patrick R. Callier (Dept. of Linguist, Stanford Univ., Bldg. 460, Stanford, CA 94305, pcallier@stanford.edu)

Previous acoustic analyses of phonation in Beijing Mandarin and closely related varieties have shown that phonatory variability in Mandarin is conditioned by lexical tone and prosody. Tone 3 (low) and tone 4 (high fall), as-well as syllables in domain-final position, have been reported to have more negative spectral tilt (particularly lower H1-H2), indicating creaky phonation. This study examines phonation in a corpus of 6752 vowels from conversational interview speech with 16 university students in the Beijing area, using four measures of spectral tilt: H1-H2, H1-A1, H1-A2, and H1-A3. The results, which diverge from findings based on laboratory speech, are possibly the first to report on spectral tilt for tone 5 (neutral tone). In mixed-effects regression models, tone 2 (high rising) has lower H1-A1, tone 3 and tone 4 have lower H1-A1 and H1-A2, and tone 5 has lower values of H1-H2 and H1-A1. Prosodic position interacts with tone. IP-final syllables for each tone exhibit higher spectral tilt on some measures than non-final syllables, though main effects for prosodic position give lower values in IP-final position for H1-A1, H1-A2, and H1-A3. These results encourage more attention to the interaction of prosodic and tonal factors in naturalistic speaking contexts when studying phonation.

3aSC30. Acoustic measures of falsetto voice. Patricia Keating (Linguist, UCLA, Linguist, Los Angeles, CA 90095-1543, keating@humnet.ucla.edu)

Falsetto (or loft) voice is known to be generally characterized by a high fundamental frequency, a spectrum with relatively few harmonics, and a relatively strong fundamental harmonic. The present study tests the hypotheses that falsetto voice differs from modal voice not only in the acoustic measure H1*-H2* (the amplitude difference between the first and second harmonics, corrected for formant frequencies and bandwidths), but also (1) in H2*-H4* (the amplitude difference between the second and fourth harmonics, corrected for formant frequencies and bandwidths), and (2) in spectral noise as measured by harmonics-to-noise measures. These and other acoustic measures will be obtained for selected pairs of vowels using VoiceSauce, the UCLA program for voice analysis. The test corpus comprises pairs of sentences taken from two readings of the story "Little Red Riding Hood" by 17 speakers of American English (8 women, 9 men). Each speaker read the story first in a neutral voice, and then using character voices for the dialogs, which for these 17 speakers were often in falsetto. The goal of the study is a reliable acoustic measure of falsetto voice.

3aSC31. Phonetic marking of stance in a collaborative-task spontaneous-speech corpus. Valerie Freeman, Gina A. Levow, and Richard Wright (Linguist, Univ. of Washington, Box 352425, Seattle, WA 98195-2425, rawright@uw.edu)

While stance-taking has been examined qualitatively within conversation and discourse analysis and modeled using text-based approaches in computational linguistics, there has been little quantification of its acoustic-phonetic correlates. One reason for this is the relative sparsity of stance-taking behavior in spontaneous conversations. Another is that stance marking is embedded into a highly variable signal that encodes many other channels of information (prosody, word entropy, audience, etc.). To address these issues, we draw on varying subfields to build a corpus of stance-dense conversation and develop methods for identification and analysis of stance-related cues in the speech signal. In the corpus, dyads are engaged in three collaborative tasks designed to elicit increasing levels of investment. In these imaginary store inventory, survival, and budget-balancing scenarios, participants solve problems, but the conversation is otherwise unscripted. Based on limited previous work (Freeman, under review) and initial findings from our corpus, we predict that stance-marking employs hyperarticulation (or lack of reduction) analogous to topic or contrast focus but where reduction would be expected in the discourse structure. Stance-taking is expected to correlate with slower speaking rates, longer stressed vowels, more expanded vowel spaces, greater pitch excursions, and greater modulation of speech signal intensity.

3aSC32. The prosody-pragmatics interface: An acoustic analysis of contrasting Spanish varieties. Ryan Platz (Spanish, Italian and Portuguese, The Penn State Univ., 150 Dorchester Ln., Bellefonte, PA 16823, ryanplatz@gmail.com)

Investigating correlations between communicative contexts and prosodic elements allows researchers to exploit the (perhaps inherent) connection on the prosody-pragmatics interface. Pitch accent models such as ToBi and RaR are helpful for a descriptive analysis, but categorization of different pitch contours can be subjective and rely on the eye of the investigator. Comparing the prosodic elements that define these models' categorization parameters would provide a better approximation of the intonation patterns across specific languages varieties and pragmatic contexts. This study investigates the prosody-pragmatics interface from an acoustic perspective in an analysis of native speakers of several Spanish varieties. Participants responded to 12 situations in which two pragmatic variables were controlled for (social distance, ranking of imposition). Responses were analyzed with a Praat script to capture pitch range, pitch span and speaking rate. Results do reveal a correlation between dialect and each suprasegmental variables, but an even stronger correlation between speaker and the contextual pragmatic variables was found, irrespective of dialect. These findings provide empirical evidence that prosodic elements of the speech signal are more relevant to the given pragmatic context rather than to speaker dialect traits, suggesting intonation patterns are constructed within a communicative context rather than within the speaker's community.

3aSC33. Using developmental data to explore frequency effects in production. Melissa M. Baese-Berk (Dept. of Linguist, 1290 Univ. of Oregon, 217 Agate Hall,Eugene, OR 97403, mbaesebe@uoregon.edu) and Katherine White (Dept. of Psych., Univ. of Waterloo, Waterloo, ON, Canada)

Low-frequency words are produced with longer duration and more extreme articulation than high-frequency words. It is unclear whether these differences are the product of online processes occurring during production, or instead result from the nature of stored exemplars. One way to disentangle these accounts is to consider changes over development. Online accounts predict that frequency effects arise due to the structure of the lexicon. Therefore, as the lexicon develops, productions should change as a function of individual words' frequencies. In contrast, exemplar accounts hypothesize that these effects occur because listeners are exposed to and store words with these acoustic properties; on these accounts, the effects should remain relatively stable across the lifespan. We asked children to name pictures that differed in frequency in both the child and adult lexicons. The pictures were in one of four categories: high frequency adult/high frequency children, low frequency adult/low frequency children, high frequency adult/low frequency children and low frequency adult/high frequency children. If children's production patterns reflect frequency in the adult lexicon, this would be evidence for an exemplar account of frequency effects; if, instead, children's patterns reflect frequency in the child lexicon, this would suggest that online processes underlie these effects.

Session 3aSP

Signal Processing in Acoustics: Intelligent Feature Selection Methods for Machine Learning Problems in Acoustics

Eric A. Dieckman, Cochair
Dept. of Appl. Sci., College of William and Mary, P.O. Box 8795, Williamsburg, VA 23187

Ning Xiang, Cochair
School of Architecture, Rensselaer Polytechnic Inst., Greene Bldg., 110 8th St., Troy, NY 12180

Invited Papers

9:00

3aSP1. Automated classification of oncoming ground vehicles using acoustic echolocation and supervised machine learning. Eric A. Dieckman (Sonalysts, Inc., 84 Nicoll St, Unit 1, New Haven, CT 06511, eric.dieckman@gmail.com) and Mark K. Hinders (Dept. of Appl. Sci., College of William and Mary, Williamsburg, VA)

In order to perform useful tasks, robots must have the ability to notice, recognize, and respond to objects in their environment. This requires the acquisition and synthesis of information from a variety of sensors. Here we focus on acoustic echolocation measurements of approaching vehicles, where an acoustic parametric array propagates an audible signal to the oncoming target and the reflected backscattered signal is recorded using the Microsoft Kinect microphone array. Although useful information about the target is hidden inside the noisy time domain measurements, the Dynamic Wavelet Fingerprint process (DWFP) is used to create a time-frequency representation of the data. Intelligent feature selection allows the creation of a small-dimensional feature vector that best differentiates between vehicle types for use in statistical pattern classification routines. Using experimentally measured data from real vehicles at 50 m, this process is able to correctly classify vehicles into one of five known classes with 94% accuracy. Fully three-dimensional simulations allow us to study the nonlinear beam propagation and interaction with real-world targets to improve classification results.

9:20

3aSP2. Use of supervised machine learning for real-time classification of underwater targets using Autonomous Underwater Vehicle sampled bistatic acoustic scattered fields. Erin M. Fischell and Henrik Schmidt (Mech. Eng., MIT, 77 Massachusetts Ave., 5-204, Cambridge, MA 02139, emf43@mit.edu)

A method has been developed for the classification of underwater target geometry using bistatic acoustic amplitude data collected by an Autonomous Underwater Vehicle (AUV) as it follows a selected path through the scattered field created by a fixed source insonifying a target. The mobility of an AUV allows it to exploit features of this field in three dimensions. The classification method presented includes offline and onboard processing components, which use a combination of signal processing, vehicle behaviors, and machine learning in the form of Support Vector Machines (SVMs) to extract target geometry from collected acoustic data. The offline training and analysis step creates training and test vector sets in a selected feature space from existing scattered field data and outputs models for target classification, confidence, and feature ranking. Several algorithms are explored for selecting the feature space used by the SVM. The models produced by the offline processing step are used in the real-time classification processing chain onboard an AUV sampling an unclassified target's scattered field. The presented simulation results use scattered fields modeled using OASES-SCATT and demonstrate real-time processing and path planning in the LAMSS MOOS-IvP simulation environment. [Work supported by ONR Code 321 OA and NSF GRFP.]

Contributed Papers

9:40

3aSP3. Learning environmentally dependent feature representations for classification of objects on or buried in the seafloor. Jason E. Summers (Appl. Res. Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com), Timothy C. Havens (Depts. of Elec. and Comput. Eng. and Comput. Sci., Michigan Technolog. Univ., Houghton, MI), and Thomas K. Meyer (Appl. Res. in Acoust. LLC, Culpeper, Virginia)

Classification performance of underwater objects is determined in large part by the acoustic response of the environment. This is a consequence of environmental clutter and of the altered response of objects that occurs through interaction with the environment—particularly for low-frequency scattering from objects on or buried in the seafloor. Therefore, to achieve robust classification, it is necessary that signal-processing algorithms account for the local context of the underwater environment. Theoretical, experimental, and numerical studies have characterized well the influence of the environment on scattering from objects for a variety of burial conditions, object shapes, structures, and materials, and sediment types and topographies. However, the relationship between these results and measurable features useful for classification remains to be fully developed. Here, we describe a machine-learning approach to selecting features for environmentally adaptive classification that seeks to algorithmically select from measured and modeled data those feature representations best suited to object

classification in varying environmental contexts. We follow a data-driven online learning approach that, while leveraging current theories and models describing environment/target interaction, does not impose an artificial limitation on the true diversity of targets and their environments. Preliminary results are presented for measured and modeled data.

9:55–10:15 Break

10:15

3aSP4. A method for deconstructing the relaxation absorption spectrums of gas mixtures. Tingting Liu, Shu Wang, and Ming Zhu (Dept. of Electronics and Information Eng., Huazhong Univ. of Sci. and Technol., Wuhan, Hubei 430074, China, liu.tt199104@gmail.com)

It is acknowledged that the relaxation absorption spectrum curves can be employed to detect gas compositions. However, it still remains a challenge to extract the composition information from the spectrums. In this paper, a method is proposed for deconstructing the relaxation absorption spectrums of a gas mixture. Based on the deconstruction results, a gas relaxation absorption spectrum is constituted by the curve tendency, which is determined by the gas effective specific heat, and the curve position, which is determined by the gas effective relaxation time. And the effective specific heat of a gas mixture is the simple addition of the compositions' independent effect, while the effective relaxation time is the complex interaction effect between different compositions. Consequently, the deconstruction method can deconstruct the relaxation absorption spectrums of a gas mixture into the simple sum of its compositions' spectrums. By the comparison between the theory curves and the experimental data, the validity of the deconstruction method is proved. The method holds great potentials for the obtainment of composition information in gas detection.

10:30

3aSP5. Selected prosody features in an accent learning system. Xiao Perdereau (Burgundy Univ., 9, Av. A. Savary, BP 47870, Dijon, France, xiao.chen-perdereau@u-bourgogne.fr)

Human speech interaction experiments have been structured as temporal and spectral representations of acoustical waveforms to better understand human facilities in handling previously unheard non-native accents of a natural language. We used Mandarin as target language for native and non-native speakers. Identical linguistic materials have been applied to both human and machine learning process. The word sequences have been selected for inference. The outcomes from human speech interactions are analyzed using computer. Speech prosody characteristics have been selected to identify and to classify naturally accented speeches. Correct and incorrect word sequences for each accented speech are discriminated. Starting with a limited stored samples of natural speech data, new signals of accented speech could be detected. Relevant acoustic features extracted from human speech are used in a comparative processing algorithm. Modified speech prosodies are produced artificially by modulating a relatively small number of physical parameters. On one hand, the variabilities regenerated by the dynamic device are fed foreword for extended human accented speech training, on the other hand, recycled acoustic signal acquisitions improve our machine knowledge source through gradual accumulation. This work is part of the development of multimedia tools integrated in an accent learning systems.

10:45

3aSP6. Buzz, squeak, and rattle noise classification by using acoustic-fingerprinting technology. Dae-Hoon Seo, Jung-Woo Choi, and Yang-Hann Kim (Mech. Eng., Korea Adv. Inst. of Sci. and Technology(KAIST), 373-1 Guseong-dong, Yuseong-gu, Daejeon 305-701, South Korea, ihuny@kaist.ac.kr)

The buzz, squeak, and rattle (BSR) noises are three representative types of the automotive interior or exterior noise. Some of BSR noises have very short duration in time, and hence, it is difficult to detect various BSR noises in low SNR situation. However, each BSR noise signal has a unique time-frequency characteristic, depending on the various contacting materials as well as the excitation forces. Therefore, it is necessary to utilize the time-frequency characteristic of the BSR noise to specify the origin of a noise source. In this paper, we propose a novel method and system for identifying BSR noises. For accurate classification of noise sources, a noise-fingerprinting and matching technique based on the pattern classification is devised. The identification test with the real BSR noise data shows that the proposed method can accurately classify the noise source even in the low SNR condition.

11:00

3aSP7. Spectrotemporal Gabor filters for feature detection. Leslie S. Smith and Andrew K. Abel (Computing Sci. and Mathematics, Univ. of Stirling, Stirling, Scotland FK9 4LA, United Kingdom, l.s.smith@cs.stir.ac.uk)

Features (landmarks) in sound are located in time and spectrum. Two dimensional (time x spectrum) Gabor filters can be used to detect useful classes of these. We use a set of logarithmically spaced bandpass filters whose outputs are coded as spikes to perform spectral analysis. These are convolved with Gabor filters to create spectrotemporal feature maps. Using auditory-nerve like spikes to code zero-crossings retains precise timings and amplitude information, and makes convolution computation relatively straightforward. Gabor patches with "horizontal" bars (parallel to time axis) can be used to detect harmonicity, and patches with "vertical" bars (parallel to spectrum axis) can detect envelope modulations. Because the time resolution is maintained in the preprocessing, vertical bars may be close together (e.g., 5 to 10 ms apart), enabling detection of amplitude modulation due to unresolved harmonics. This is useful for both for speech voicing detection, and for animal utterances. Such filters may be localized in spectrum, allowing tracking of voicing energy. Filters with bars at other angles can detect frequency modulation. Using constellations of these features (and others, such as onsets), we can characterize and interpret sound sources.

Session 3aUWa

Underwater Acoustics: Acoustic Signal and Noise Propagation and Scattering

Kathleen E. Wage, Chair
George Mason Univ., 4400 University Dr., Fairfax, VA 22030

Contributed Papers

8:00

3aUWa1. Acoustic scattering from a sand layer and rock substrate with rough interfaces using the finite element method. Anthony L. Bonomo, Marcia J. Isakson, and Nicholas P. Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, anthony.bonomo@gmail.com)

The finite element method is used to study the acoustic scattering from a layer of sand overlying a rock substrate. All the modeling is done in two dimensions. Both the water-sand interface and the sand-rock interface are modeled as random rough surfaces following a modified power law spectrum. The rock substrate is assumed to be an elastic solid. Three sediment models are used for the sand layer: the full Biot model for poroelastic media, an effective density fluid model based on the Biot model, and a simple fluid model. The effect of the choice of sediment model used for sand is studied. The finite element results are also compared with perturbation theory and the Kirchhoff approximation in order to further evaluate the validity of considering the underlying interfaces to be flat as a rough sand-rock interface cannot be handled by these models. [Work supported by ONR, Ocean Acoustics.]

8:15

3aUWa2. Modeling range dependent sediment and interface roughness effects on propagation loss with finite elements. Marcia Isakson and Nicholas Chotiros (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78713, misakson@arlut.utexas.edu)

At the Target and Reverberation Experiment (TREX) off the coast of the Florida panhandle in May 2013, propagation loss was measured over a track with significant sediment and interface roughness variability. In this study, the finite element method is applied to model the effects of sediment and interface roughness variability on transmission loss. Finite elements provide a full wave solution to the Helmholtz equation to the accuracy of the discretization. Therefore, it provides both forward and backward propagation fields. Where available, data will be taken from TREX environment measurements. Additionally, the results will be compared with an energy loss model which relies on the product of the range dependent reflection coefficient. [Work supported by ONR, Ocean Acoustics.]

8:30

3aUWa3. Scattering of sound by a cylindrically symmetric seamount. Ronald Pannatoni (540 Mark Dowdle Rd., Franklin, NC 28734, elliptic@alum.mit.edu)

A three-dimensional waveguide is used to model scattering of sound by a cylindrically symmetric seamount. The waveguide is "closed" in the sense that it has a boundary below the ocean bottom. This makes the spectrum of local wavenumbers discrete. Partial differential equations govern the modal expansion coefficients of the acoustic pressure and of the radial pressure gradient. Their solutions must be finite-valued at the axis of symmetry and satisfy outgoing radiation conditions at the perimeter of the base of the seamount. Direct numerical integration of this problem is unstable, but with variation of parameters and a Riccati transformation an equivalent problem is obtained for which numerical integration is stable. This approach is compared with an alternative [Pannatoni, POMA **14**, 070003 (2011)] that uses leaky modes of an "open" waveguide having no boundary below the ocean bottom. It is shown how coupling between nearest-neighbor modes of the "closed" waveguide relates to the leaky modes of the "open" waveguide. This coupling restricts the step size that numerical integration can use.

8:45

3aUWa4. Computation and accuracy trade-offs in applied reverberation modeling. Anthony I. Eller (OASIS, Inc., 1927 Byrd Rd., Vienna, VA 22182, ellera@oasislex.com) and Kevin D. Heaney (OASIS, Inc., Fairfax Station, VA)

Reverberation modeling and prediction are a research area of intense interest, often involving large, high resolution databases and advanced numerical techniques, and stretching modern computers to their limits. Nevertheless, applications of underwater acoustic systems often need estimates of reverberation sooner than they can be provided by the research community. What results is that applied models for reverberation, in order to meet the pressing needs with available computer memory and allowable computation times, often employ approximations or shortcuts that fall short of current research modeling standards. This paper traces several of the approximations used in reverberation models in the past in order to meet system design and deployment needs. The results show an inherent conflict between basic research and applied interests. The traditional DOD paradigm is that basic research results transition to the applied world. History suggests the opposite, however: Applied research ignores the torpedoes and moves full speed ahead where needed, then forges the chain used to pull reluctant basic research in the needed direction.

9:00

3aUWa5. The research on the coverage area of multistatic sonar under various work modes. Xueli Sheng (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, Heilongjiang, China), Jia Lu (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin Eng. University Shuisheng Bldg. 803, Nantong St. 145, Harbin, Hei Longjiang 150001, China, lujia0507@163.com), Weijia Dong, Jingwei Yin, Longxiang Guo, and Xiaochen Wu (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, Heilongjiang, China)

Absorption loss, which is an important characteristic in the field of underwater acoustic, caused by sea water is often neglected in the research on the coverage area of bistatic sonar. In this article, the coverage area of bistatic sonar considering absorption loss is studied. The change of coverage area caused by absorption loss and the approximation are discussed. When multistatic sonar works in different modes, detection criterion varies, and the coverage area cannot be simply considered as the union set or the intersection of coverage area of each bistatic sonar unit. The definition of multistatic coverage area is discussed, and the coverage area respect to the most relaxed criterion and the strictest criterion is studied.

9:15

3aUWa6. European project for achieving quieter oceans by shipping noise footprint reduction. Christian Audoly and Céline Rousset (CEMIS/AC, DCNS Res., Le Mourillon, BP 403, Toulon 83055, France, christian.audoly@wanadoo.fr)

There is a growing consensus among the scientific community for the need to mitigate underwater noise footprint due to shipping, in order to prevent negative consequences to marine life. In that context, AQUO project started in October 2012, in the scope of the FP7 European Research Framework, for three years duration. The final goal is to provide policy makers with practical guidelines and solutions, acceptable by shipyards and ship owners, in order to mitigate underwater noise due to noise radiation from ships. First, a general presentation of the project will be given, and the logical relationship between the tasks will be outlined. A key element is the development of a "Shipping noise footprint assessment tool," derived from the software Quonops©. This tool can predict the noise map in a maritime area using real time information on ship traffic. Many other studies are undergoing, covering ship underwater radiated noise, propeller noise (including cavitation), and bio-acoustic experiments. The AQUO project team is composed of ship industry, specialized companies, a classification society, research centres and academics, allowing addressing the complexity of the topic. In a second part, some recent results from the project will be given. These will include, for example, the definition of indicators for the impact of shipping noise on marine life, results from vibro-acoustic measurements on several ships at sea, and results from bio-acoustic studies.

9:30

3aUWa7. Mode analysis of distant shipping noise in the SPICE experiment. Mehdi Farrokhrooz, Kathleen E. Wage (Elec. and Comput. Eng., George Mason Univ., 4450 Rivanna River Way PMB3740, Fairfax, VA 22030, mfarrokh@masonlive.gmu.edu), Matthew A. Dzieciuch, and Peter F. Worcester (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

A major source of ambient noise for frequencies below 100 Hz is due to distant shipping (Wenz, 1962). Shipping noise is trapped in the SOFAR channel and propagates long distances. Since this part of ambient noise excites the low normal modes, it decays with increasing depth. A modal analysis of shipping noise provides valuable insights about its depth dependence. SPICEX offered a unique opportunity to study the ambient noise in the Northeastern Pacific using two vertical line arrays an axial array and a deep array. A 40-hydrophone axial array spanned the SOFAR channel and recorded ambient noise at regular intervals over a year. The axial array data makes it possible to estimate the wavenumber-frequency spectra of ambient noise. We can estimate the mode powers associated with the angle-limited noise in the wavenumber-frequency spectra using a least squares fit. Given the estimated mode powers, depth-dependence of the shipping noise component is predictable. This talk compares the predicted noise levels as a function of depth with data measured by the deep SPICEX line array.

9:45

3aUWa8. Broadband incoherent virtual array beamforming on small scale array for passive detection. Weijia Dong, Xueli Sheng, Chunyan Sun (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., No.145 NanTong St., Harbin 150001, China, messi_881003@163.com), Jia Lu, and Hu Shen (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, Hei Longjiang, China)

It cannot be guaranteed that array spacing corresponding to half-wavelength of each frequency during broadband signals passive detection process. Therefore, for high-frequency, side lobe will rise when array spacing is half the wavelength of the low-frequency. In addition, the main lobe of small-scale array will be wider due to limited array aperture. In order to resolve these two disadvantages for passive detection of small-scale array, virtual array beamforming based on time-delay information is studied in this paper. But this technology often requires high sampling frequency which will increase computational complexity. In order to use virtual array beamforming to processing broadband signals at lower sampling frequency, broadband incoherent virtual array technology based on phase information is introduced in this paper. The effectiveness of these two methods is shown by simulation and sea experiment.

10:00–10:15 Break

10:15

3aUWa9. Alternatives to traditional probabilistic approaches for incorporating environmental uncertainty in ocean acoustics. Steven I. Finette (Acoust. Div., Naval Res. Lab., 4555 Overlook Ave. SW, Washington, DC 20375-5320, steven.finette@nrl.navy.mil)

Quantifying environmental uncertainty and its corresponding mapping onto the acoustic field is a subject of considerable importance for underwater acoustics modeling and simulation. The ability to make accurate numerical predictions of basic acoustic quantities, as well as sonar system performance in complex ocean environments depends, in part, on objectively quantifying our imperfect knowledge of the waveguide properties. Over the past few years, a number of approaches have been considered to quantify uncertainty in ocean-acoustics modeling; these methods are grounded in some form of traditional probabilistic reasoning. While probability theory provides a natural framework for representing and quantifying environmental uncertainty, a significant amount of information must be available to specify density functions for the parameters and fields so that the corresponding acoustic field statistics can be computed. This talk will overview alternatives to traditional probability-based reasoning that have been introduced in research areas where conventional specification of probability distributions are not a practical means for describing uncertainty. [Work supported by base funding from ONR.]

10:30

3aUWa10. Spatial uncertainty in higher fidelity acoustic models, and impact on model-data comparisons. Richard L. Campbell, Kevin D. Heaney (OASIS Inc, 11006 Clara Barton, Fairfax Station, VA 22039, campbell@oasislex.com), Phil Abbot, and Chris J. Emerson (OASIS Inc, Lexington, MA)

Acoustic propagation models, such as the parabolic equation (PE), are typically parameterized by a single source depth and one or more receiver depths and ranges. At the higher frequencies, propagation angles, and ranges enabled by advances in computer hardware and refinement of algorithms, the acoustic field predicted by the model can contain features at considerably finer scale than the resolution of the requested output points. The typical practice of simply sampling the field at the output points can introduce artificial large scale structure in the output, due to aliasing. However, averaging out the fine-scale spatial variability can also result in a misleadingly precise value for transmission loss (TL) at a given point. We show that reconciliation of acoustic model predictions with experimental data can be greatly improved by presenting the TL variability due to the expected uncertainty of source and receiver position within an otherwise deterministic model. Even with good estimates of source and receiver positions, this variability can be greater than that due to uncertainty in the model's environmental inputs. Model-data comparisons are demonstrated with examples from a recent ONR effort in the eastern Gulf of Mexico which included controlled TL measurements at 0.4–2 kHz in shallow water.

10:45

3aUWa11. Correlation matrices as a method of source function isolation and classification in ambient noise. Stephen Nichols (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg., State College, PA 16802, smn5198@psu.edu) and David L. Bradley (Appl. Res. Lab., The Penn State Univ., State College, PA)

At low frequencies (1–100 Hz), the deep-ocean ambient noise field is a mixture of many source types, including seismic activity, marine life, ship traffic, seismic airgun surveys, and nonlinear surface wave interactions. The location of the sensors used to monitor these types of noise typically result in an extremely dynamic recorded ambient noise field. This study proposes the use of correlation matrices as a tool for identifying the predominant noise sources present in samples of ambient noise. These correlation matrices work by identifying frequency ranges in which the noise levels tend to change at the same time, thus identifying frequency ranges where noise is driven by one particular source mechanism. The effectiveness of this strategy will be demonstrated using low-frequency deep-ocean ambient noise recorded in the Pacific, Atlantic, and Indian Oceans by the Comprehensive

Nuclear-Test Ban Treaty Organization (CTBTO) hydroacoustic monitoring system.

11:00

3aUWa12. Overview of the Target and Reverberation EXperiment 2013 (TREX13). Dajun Tang and Brian T. Hefner (Appl. Phys. Lab, Univ of Washington, 1013 NE 40th St., Seattle, WA 98105, djtang@apl.washington.edu)

The Target and Reverberation Experiment (TREX13) is a shallow water reverberation experiment that endeavored to measure contemporaneously acoustic and adequate environmental data so detailed model/data comparison can be achieved and important environmental factors can be identified for different applications. TREX13 was sponsored by ONR Ocean Acoustics and the SERDP DoD programs. It was conducted during April to June of 2013 off the coast of Panama City, Florida, with participation from multiple institutions and involving three research vessels: The R/V Sharp and R/V Walton Smith from the United States, and the Canadian Force Auxiliary Vessel Quest. From a SONAR viewpoint, reverberation consists of two-way propagation and a single backscatter. Therefore, reverberation, transmission loss, and bottom backscatter were repeatedly measured over a time period of several weeks in the frequency band of 2–10 kHz, along with extensive environmental measurements. Discussed will be planning and execution of the field experiments, strategies and steps for data analysis, and modeling efforts.

11:15

3aUWa13. Solving the inverse problem to extract ship radiated spectra in ocean environments. David P. Knobles (ARL:UT, Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78757, knobles@arlut.utexas.edu)

In ocean acoustics, one is often asked to find the Green's function G from the measured response R in R = GS, where S is the source spectrum. Experiments are often conducted where R is measured and S is known. However, the problem becomes more complicated when S is not known. The application of interest here is the solution for the source spectrum S = Inverse(G) R for a fast moving surface ship in uniform motion in an ocean environment where the propagation includes the interaction of sound with the seabed. What makes the solution possible is the multimodal nature of the propagation over a broadband of frequencies. As an application, the received spectra of merchant ships inside the Reliable Acoustic Path (RAP) range were recorded in the North Pacific Ocean on a vertical line array in an experiment called Church Opal in 1975. The effects of the seabed are clearly evident on the signals received on a hydrophone below the reciprocal depth. As a means of validation, the source spectra solutions of five merchant ships are compared to the spectra of similar ships recorded in other deep-water environments where the effects of the seabed are not present.

11:30

3aUWa14. Utilizing an extended target for high frequency multi-beam sonar intensity calibration. John L. Heaton (Ocean/Mech. Eng., Univ. of New Hampshire, 24 Colovos Rd., Durham, NH 03824, jheaton@ccom.unh.edu), Thomas Weber (Ocean/Mech. Eng., Ctr. for Coastal and Ocean Mapping/Joint Hydrographic Ctr., Univ. of New Hampshire, Durham, NH), Glen Rice (NOAA Office of Coast Survey, Ctr. for Coastal and Ocean Mapping/Joint Hydrographic Ctr., Univ. of New Hampshire, Durham, NH), and Xavier Lurton (IMN/NSE/AS, IFREMER, Plouzane, France)

There is an interest in expediting intensity calibration procedures for Multi-Beam Echo-Sounders (MBES) to be used for acoustic backscatter measurements. To this end, a target was constructed of irregularly oriented small chain links arranged in a "curtain" simulating an extended scattering surface, such as the seafloor. Tests with a 200-kHz, 7°, SIMRAD EK60 Split-Beam Echo-Sounder (SBES) were performed in a tank in order to investigate the targets angular- and range-dependent scattering strength. These tests suggest that the amplitude envelope of the scattered signal is Rayleigh distributed and that the backscatter strength depends linearly on the number of active scattering elements. Following these initial validation tests, a 200 kHz Reson SeaBat 7125-SV2 MBES was calibrated in the tank with the same extended target. During the calibration, the MBES was rotated so that every beam was incident on the target. This calibration test was performed once when the target was at normal and once at oblique (45°) incidence. The final output is a "catch all" beam-dependent calibration coefficient, C, determined from the sonar equation.

11:45

3aUWa15. Development of a semi-analytical model for the underwater radiated noise from a driven pile—Comparison of the stationary phase approximation with exact integration for computing an inverse Fourier Transform of vertical wavenumber. Marshall V. Hall (Marshall Hall Acoust., 9 Moya Crescent, Kingsgrove, NSW 2208, Australia, marshallhall@optushome.com.au)

A semi-analytical model is being developed for the noise radiated underwater when a cylindrical pile is struck axially by a hammer. The model is based on the coupled equations of motion for axial and radial vibration of a thin shell. It yields frequency-dependent axial phase velocity and attenuation (due to radiation) and produces a complete description of the shell vibration. For the purpose of describing the total radiated sound pressure, a harmonic solution is obtainable. The "Transform formulation of the pressure field of cylindrical radiators" by Junger and Feit is adopted. The result includes an inverse Fourier transform of a function of vertical wavenumber, and when the integrand is simplified using the stationary-phase approximation, the resulting pressure at short ranges is very large along a downward line in the range-depth plane. The slope of this line is that of the "Mach waves" described by Reinhall and Dahl, and the pressure is many times as great as that of the Mach waves. This approximation is therefore inapplicable. Results are also presented based on using wavenumber integration to compute the Fourier transform.

12:00

3aUWa16. Approximation of a physics-based model of surface reflection loss. Adrian D. Jones, Alex Zinoviev, and David W. Bartel (Maritime Div., DSTO, P.O. Box 1500, Edinburgh, SA 5111, Australia, bearjones@adam.com.au)

A model of coherent acoustic reflection loss at the ocean surface had been prepared by the authors by combining a model of surface roughness loss with a description of surface incidence angle which accounted for the refractive effects of a uniformly stratified distribution of wind-driven bubbles. Here, surface roughness loss was based on a second-order small-slope approximation, and the surface incidence angle was obtained using a formulation for stratified media from Brekhovskikh applied to the sound speed variation resulting from the bubble distribution used by Ainslie [JASA **118**, (2005)]. More recent work by the authors has shown that the analyses for each of surface roughness loss, and surface incidence angle, may be approximated adequately by relatively simple expressions, and that the complete model of surface reflection loss inclusive of the refractive effects of bubbles may be approximated in expressions suitable for hand calculations. Results from the use of this approximated model with a Gaussian-beam acoustic propagation code are compared with results obtained from the authors' more complete model for several surface ducted transmission scenarios. Both sets of results are also compared with predictions based on Monte Carlo parabolic equation (PE) transmission calculations.

Session 3aUWb

Underwater Acoustics and Acoustical Oceanography: Underwater Acoustics and Acoustical Oceanography Poster Session

Gopu R. Potty, Chair
Dept. of Ocean Eng., Univ. of Rhode Island, 115 Middleton Bldg., Narragansett, RI 02882

All posters will be on display from 9:00 a.m. to 12:00 noon, To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 9:00 a.m. to 10.30 a.m. and contributors of even-numbered papers will be at their posters from 10:30 a.m. to 12:00 noon.

Contributed Papers

3aUWb1. On the inversion of sediment density profile by the image source method. Achraf Drira, Laurent Guillon, and Abdel-Ouahab Boudraa (Ecole navale/IRENav, Ecole navale/IRENav, BCRM Brest, CC600, Brest cedex 9 29240, France, achraf.drira@ecole-navale.fr)

Image Source Method is a recently developed method for geoacoustic inversion. Under the Born approximation, the reflection of a spherical wave above a stratified seafloor is modeled as the simultaneous emission of image sources, symmetric of the real one relatively to the layers. These image sources are detected and located by the use of Teager-Kaiser Energy Operator (TKEO), which amplifies sudden changes in signal amplitudes, and by a triangulation scheme. Time and arrival angle of the recorded signals coming from the image sources are the input values of the inversion algorithm which gives the thickness and sound speed of each detected sediment layer by the use of Snell-Descartes laws. The objective of the present work is to extend this method to the inversion of the sediment density profile. To this end, one supplementary data is required. Experimental amplitudes corresponding to arrival times of the reflected signals, detected by TKEO, are compared to theoretical ones computed by a numerical evaluation of the Sommerfeld integral. Having first inverted sound speed profile and neglecting absorption coefficient, density is the only remaining parameter and thus can be obtained. The effectiveness of this method is tested on both synthetic and real data.

3aUWb2. Statistical inference of seabed sound-speed structure in the Gulf of Oman Basin. Jason D. Sagers and David P. Knobles (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, sagers@arlut.utexas.edu)

Addressed is the statistical inference of the sound-speed depth profile of a thick soft seabed from broadband sound propagation data recorded in the Gulf of Oman Basin in 1977. The acoustic data are in the form of time series signals recorded on a sparse vertical line array and generated by explosive sources deployed along a 280 km track. The acoustic data offer a unique opportunity to study a deep-water bottom-limited thickly sedimented environment because of the large number of time series measurements, very low seabed attenuation, and auxiliary measurements. A maximum entropy method is employed to obtain a conditional posterior probability distribution (PPD) for the sound-speed ratio and the near-surface sound-speed gradient. The multiple data samples allow for a determination of the average error constraint value required to uniquely specify the PPD for each data sample. Two complicating features of the statistical inference study are addressed: (1) the need to develop a cost function that can both utilize the measured arrival structure and mitigate the effects of data errors and (2) the effect of small bathymetric slopes on the structure of the bottom interacting arrivals. [Work supported by ONR.]

3aUWb3. Issues in estimating the seafloor scattering cross section with synthetic aperture sonar. Derek R. Olson (Acoust., The Penn State Univ., 201 Appl. Sci. Bldg, University Park, PA 16802, olson.derek.r@gmail.com) and Anthony P. Lyons (Appl. Res. Lab., Penn State, University Park, PA)

The high resolution capabilities of synthetic aperture sonar (SAS) make it an attractive technology for estimating the seafloor scattering cross section. However, differences between traditional systems used for backscattering measurements, whose apertures span a few wavelengths, and synthetic arrays, whose apertures can span thousands of wavelengths, can violate the assumptions used in typical cross section estimation techniques. The issues introduced by very long synthetic arrays are caused by the integration of acoustic energy over a range of azimuth and grazing angles in the local coordinates of the resolved area on the seafloor. The resulting pixel intensity is not directly proportional to the scattering cross section, and methods other than solving the sonar equation are required. This research explores the effect of naively using SAS data to estimate the cross section and presents alternative estimation techniques.

3aUWb4. Estimation of seabed bottom loss using wideband signal for underwater acoustic communication. Sung-Hoon Byun, Sea-Moon Kim, and Yong-Kon Lim (Ocean System Eng. Res. Div., KRISO, 171 Jang-dong Yeseong-gu, Daejeon 305-343, South Korea, byunsh@gmail.com)

Performance of acoustic communication through shallow underwater channel is affected by surface and bottom interaction, and the bottom loss has great importance for long-range signal transmission. In 2013, KRISO has performed a shallow-water communication experiment near Jeju island, South Korea, and the experiment aims at evaluating communication link budget as well as measuring the channel fading statistics. The measured sound velocity profiles show the existence of strong thermocline resulting in downward refraction and therefore its long-range signal propagation is strongly dependent on the seabed characteristics. We use a wideband signal, which was originally designed for channel impulse response estimation. The signals were recorded using a four-element receiver array with two different source depth configurations which enable observing the seabed at various grazing angles. The estimated bottom loss is used to estimate the transmission losses at longer ranges, and they are compared with the measured values to examine the estimated bottom loss.

3aUWb5. Simulation and analysis of acoustic impedance measurement techniques for marine sediments using sonar. João P. Ristow (Mech. Eng., Federal Univ. of Santa Catarina, Rua Lauro Linhares, 657, Apto. 203B, Florianápolis, Santa Catarina 88036-001, Brazil, jpristow@gmail.com), Guillaume Barrault (Oceanogr., Federal Univ. of Santa Catarina, Florianápolis, Santa Catarina, Brazil), Julio A. Cordioli, Gregório G. Azevedo (Mech. Eng., Federal Univ. of Santa Catarina, Florianápolis, Santa Catarina, Brazil), Antônio H. Klein, and Marina Bousfield (Oceanogr., Federal Univ. of Santa Catarina, Florianápolis, Santa Catarina, Brazil)

The characterization of seabed sediments using acoustics techniques is a wide field of research. A key feature in the sediment characterization is the estimation of its acoustic impedance, from which density and wave propagation speed in the sediment can be derived. The sea bottom acoustic impedance is a property that rules the backscattering strength, together with other sediment features such as slope and roughness. In this sense, sonar devices can be used to measure the backscattering strength of a material lay up on the sea bottom and estimate the aforementioned sediment features. However, one of the main difficulties is to obtain sufficient complementary information in order to decompose the backscattering strength of a giving sea bottom and obtain its proper acoustic impedance. Hence, the aim of this work is to evaluate and compare different techniques for measurement and signal processing that will be used to indirectly calculate the sediment's acoustic impedance from numerically simulated sonar data.

3aUWb6. Comparison between the Biot-Stoll model, the grain-shearing model, and the effective density fluid model with respect to sediments in the Yellow Sea. Eunghwy Noh, Hunki Lee (School of Mech. Eng., Yonsei Univ., Eng. Bldg. A117, 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, hwyya@yonsei.ac.kr), Oh-Cho Kwon (The 6th R&D Inst. Naval System Directorate Principal Researcher, Agency for Defense Development, Changwon, South Korea), and Won-Suk Ohm (Mech. Eng., Yonsei Univ., Seoul, South Korea)

There are some distinctive geophysical properties of sediments in the Yellow Sea, where mudflats are developed extensively. To select a physical model for acoustic wave propagation in the sediments is a critical issue in the analysis and the conceptual design of buried object scanning sonars. We consider three different sediment models to describe physical and geoacoustic properties of the sediments in the Yellow Sea: the Biot-Stoll model, the grain-shearing model, and the effective density fluid model (EDFM). These sediment models are evaluated in terms of their predictions of sediment properties compared with the measurement data. [This work was conducted in the Unmanned Technology Research Center (UTRC) sponsored by the Defense Acquisition Program Administration (DAPA) and the Agency for Defense Development (ADD) in the Republic of Korea.]

3aUWb7. Propagation of nonlinear sound beams in marine sediments. Hunki Lee, Eunghwy Noh (Mech. Eng., Yonsei Univ., 50 Yonsei-ro, Seodaemun-gu, Seoul 120-749, South Korea, ssevi2@yonsei.ac.kr), Oh-Cho Kwon (The 6th R&D Inst. Naval System Directorate Principal Researcher, Agency for Defense Development, Changwon, South Korea), and Won-Suk Ohm (Mech. Eng., Yonsei Univ., Seoul, South Korea)

Parametric arrays are recently considered as directional sound sources for locating buried objects in marine sediments. To understand how nonlinear sound beams generated by parametric arrays propagate in marine sediments, a theoretical framework that accounts for the combined effects of diffraction, absorption, nonlinearity, and the physics of sediment is necessary. In this paper, the KZK equation is extended to augment a number of well-known sediment models. Nonlinear evolution of sound beams are computed and compared under different sediment models, namely, the fluid model, Buckingham's grain-shearing model, the Biot-Stoll model, and the effective density fluid model (EDFM). [This work was conducted in the Unmanned Technology Research Center (UTRC) sponsored by the Defense Acquisition Program Administration (DAPA) and the Agency for Defense Development (ADD) in the Republic of Korea.]

3aUWb8. Broadband modeling of sound propagation in shallow water with an irregular elastic bottom. Li Li (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Shuisheng Bldg. Rm. 1303, Nantong St., No.145, Nangang District, Harbin, Heilongjiang 150001, China, liliandxy@126.com), Shengchun Piao, Haigang Zhang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China), Yaxiao Mo, and Jun Tang (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, China)

The purpose of this work is to study the propagation of broadband sound pulses in shallow water environments. It is essential for an underwater pulse propagation model to handle bottom interaction, range-dependence, and wide-angle propagation in shallow water. Therefore, a comparatively realistic model that consists of a fluid overlying an irregular elastic bottom is treated, where the effects of shear wave are included. The range-dependent seismo-acoustics problem in frequency-domain is solved by a parabolic equation model. Fourier synthesis of frequency-domain solutions is implemented to model the received time series of a broadband sound propagation. And parallel programming is tried to improve the computational efficiency. Dispersion characteristics are exhibited by multiple mode arrivals during the propagation, including the dispersion of normal modes and mode 0 (the Scholte wave). The dispersion analysis of normal modes and the Scholte wave are demonstrated under different types of elastic bottoms. Energy converting between trapped modes and leaky modes due to slope of bottom is also analyzed.

3aUWb9. Laboratory measurements of high-frequency, broadband acoustic scattering of growing sea ice and oil beneath sea ice. Christopher Bassett, Andone C. Lavery, and Ted Maksym (Appl. Ocean Phys. & Eng., Woods Hole Oceanographic Inst., 266 Woods Hole Rd., MS# 12, Woods Hole, MA 02543, cbassett@whoi.edu)

The morphology of sea ice during the early stages of growth is strongly dependent on environmental conditions. Under calm conditions, congelation ice forms through downward growth of ice crystals from the water surface. Under turbulent conditions (surface waves), rapid freezing of ice crystals occurs in the upper water column (frazil ice), eventually consolidating into pancake ice through repeated collisions and agglomeration of the loose frazil crystals. It is expected that high-frequency scattering from the basal layer of the ice varies for different sea ice types and can reveal structural information that governs the behavior of the ice and its interactions with the environment. Broadband scattering measurements of sea ice are presented beginning with ice-free conditions and through initial stages of growth in laboratory experiments for both congelation and frazil ice. With increased interest in drilling for hydrocarbon resources in the Arctic and the associated environmental concerns of an oil spill in ice-covered waters, improved methods for detection of crude oil both under or frozen within sea ice are needed. Acoustic scattering data are presented demonstrating how the scattering changes when crude oil is spilled beneath the ice.

3aUWb10. Detectability of navigation obstacles with forward looking sonar in presence of boat wakes: Observations in Narragansett Bay. Alexander M. Yakubovskiy, Nabin S. Sharma, and Matthew J. Zimmerman (Signal Processing, Farsounder, Inc., 151 Lavan St., Warwick, RI 02888, alex.yakubovskiy@farsounder.com)

A Forward Looking Sonar (FLS) is designed to detect obstacles ahead of a vessel. Detection performance of an FLS is affected by several environmental factors surrounding the sonar. Wakes generated by nearby vessels degrade the detection capability of an FLS. With more than 40 000 registered boats, Rhode Island's Narragansett Bay is a good example of a zone with heavy boat traffic where the performance of an FLS may be effected. FarSounder has collected vast amounts of data while testing and demonstrating its FLS sonars in Narragansett Bay. Using that data, this paper presents observations of FLS performance with and without boat wakes. In order to explain the sonar performance, a probability of wake presence in sonar field of view is estimated based on shipping density, wake size, and lifetime.

3aUWb11. Parameter measurement research of sonar echo highlights. Cheng He, Anbang Zhao, Bin Zhou, Xuejing Song, and Fang Niu (College of Underwater Acoust., Harbin Eng. Univ., Heilongjiang Province,Harbin City, Area Nangang, Nantong St. No.145, Shuisheng Bldg., Harbin, Heilongjiang 150001, China, hecheng@hrbeu.edu.cn)

Highlights sonar model provides an important basis for complex target modeling. In this paper, a parameters measurement method for the far-field sonar target highlight is proposed, time reversal mirror (TRM) technology and transponder technology combined with this method. First, get the number of target highlights. Then, put transponders with the same number along underwater the target which is measured, those transponders must be close at each highlight. A wideband signal is transmitted at measuring position which is far away from target, those transponders at target side answer one by one. Record the echoes coming from target and transponders. Time-delay, amplitude, and phase-jump of highlights can be calculated by processing those echoes in TRM and matrix operations methods. It is found through simulation by MATLAB that this target highlight algorithm can accurately measure the three important parameters of highlight model in negative SNR. By getting those parameters accurately, some other parameters can be calculated, for example, target strength. Target echo can also be predicted with those parameters, it will be very useful in target strength measurement and active target stealth technology then.

3aUWb12. The instantaneous frequency variance feature of underwater bottom target. Xiukun Li and Zhi Xia (College of Underwater Acoust. Eng., Harbin Eng. Univ., No.145, Nantong St., Rm. 305, Shuisheng Bldg., Nangang District, Harbin, Heilongjiang 150001, China, lixiukun@hrbeu.edu.cn)

The key of underwater target recognition is extracting stable target feature from the complicated mixed signal. Deal with LFM pulse, the time-frequency character of target echo is fixed and linear; however, the time-frequency feature of target echo is sensitive with the SRR, the angle of incidence wave, and the shape and the material of target. Based on the geometric acoustic scattering character of a classical bottom target model, the time-frequency distribution of target echo is treated as a bidimensional image in this paper. During with image rotation, the time-frequency distribution of target echo is transformed to a line spectrum, and then, the instantaneous frequency variance feature is extracted. The experiment data processing result has shown that the clusters of target echo and reverberation on this feature are stable for various environments and the shape and the material of targets. The method proposed in this paper is meaningful for construct a universal feature space in the detection of underwater bottom target.

3aUWb13. Spatial diversity in application of the decomposition of the time reversal operator (DORT) method to imaging of an extended target. Chunxiao Li, Mingfei Guo (MOE Key Lab. of Mech. Manufacture and Automation,College of Mech. Eng., Zhejiang Univ. of Technol., 18# Chao-Wang Rd., Zhejiang, Hangzhou 310014, China, chunxiaoli@zju.edu.cn), and Qianliu Cheng (Hangzhou Appl. Acoust. Res., Hangzhou, China)

Taking advantage of the multipath propagation in shallow water waveguide, high resolution can be achieved with the decomposition of the time reversal operator method. In this paper, we investigate how to use the spatial diversity for imaging of an extended target. It is assumed that the target is composed of an infinite number of random, isotropic, and independent scatterers, uniformly distributed over an unknown region. The eigenvalues and eigenvectors associated with the target are first determined. We then employ the frequency characterizations of scatterers from a singular value decomposition of the time reversal operator to localize and image the target. Tank experiments are carried out for a steel cylinder in two kinds of random media: a shallow water waveguide with multiple scatterers and without scatterers. The effects of the random media on the choice of the parameters of the signal processing are also studied.

3aUWb14. A possibility to use respiratory noises for detecting diver and monitoring his physiologic status. Sergei Gorovoy (Far Eastern Federal Univ., Vladivostok, Russian Federation), Vladimir Korenbaum, Aleksandr Tagiltcev, Anatoly Kostiv (Pacific Oceanologic Inst., 43, Baltiiskaya Str., Vladivostok 690041, Russian Federation, v-kor@poi.dvo.ru), Aleksey Borodin (Appl. Problems Section FEB RAS, Vladivostok, Russian Federation), Irina Pochekutova, Aleksey Vasilistov, Aleksandr Krupenkov (Pacific Oceanologic Inst., Vladivostok, Russian Federation), Denis Vlasov (Far Eastern Federal Univ., Vladivostok, Russian Federation), and Anton Shiryaev (Pacific Oceanologic Inst., Vladivostok, Russian Federation)

The objective is acoustic detection of submerged diver and monitoring his physiologic condition. A possibility to use diver's own respiratory noises for this aim is analyzed. Respiratory noises were recorded above trachea of submerged scuba diver and in the water layer of shallow-water bay. Both signals contain quasi-periodic components induced by amplitude modulation of wideband respiratory noises with the rate of breathing maneuvers. These components are detected in water layer by means of energy processing of single hydrophone response (in a frequency band of 200–500 Hz) at the distances up to 50 m. The breathing rate is estimated by means of spectral transform of the signal envelope. It is well known that this index represents human individual physiologic status. Thus, it is pertinent for submerged diver's condition monitoring. The quasi-periodic components if detected may be used to estimate time delays in responses of several hydrophones, for example, by means of the signal envelopes correlation processing. These time delays are pertinent to find diver's location by triangulation technique.

3aUWb15. Active acoustic detection of subsea oil and gas leaks; model prediction and measurements. Geir Pedersen, Rune Hauge (Instrumentation, Christian Michelsen Res. AS, P.O. Box 6031, Bergen 5892, Norway, geir.pedersen@cmr.no), and Rokas Kubilius (Ecosystem Acoust., Inst. of Marine Res., Bergen, Norway)

There is a need in the oil and gas industry for technology for real time monitoring of subsea structures. National research strategies emphasize the need to develop innovative solutions to detect, contain, and clean up spills. The technology should ideally cover a large area and be able to detect leaks up to 500 m from the sensor. As part of a project where a leak detection technology based on active acoustics is developed, effective medium models have been used to predict the acoustic signal from small oil and gas leaks. Model results were compared with multifrequency data from controlled *in situ* measurements of oil and gas releases, collected with scientific echosounders. The measurements also include stereo camera images for bubble/droplet and plume size estimation. The models generally predict the measurements within 3 dB, when taking into account the variability in plume dimensions and bubble sizes the difference is smaller.

3aUWb16. On marine mammal acoustic detection performance bounds. Yin Xian and Loren Nolte (Elec. and Comput. Eng., Duke Univ., 857 Louise Circle, Durham, NC 27705, poline3939@gmail.com)

Most current research on marine mammal acoustic detection, classification, and localization does not consider optimizing detector algorithm performance for marine mammal vocalizations that have been propagated through an uncertain multipath environment. In this study, a Bayesian likelihood ratio approach, using the original time series data, is used to benchmark optimal detection performance (ROC's) for a known marine mammal source vocalization propagated through both a known and an example uncertain ocean environment. In addition, for these same ocean environments, the detection performance (ROC's) is obtained for several algorithms based on the spectrogram of the original data. Since the spectrogram does not preserve detailed phase information contained in the original data, any algorithm based on the spectrogram is not likely to be optimum for detection. The initial results show the additional detection performance gain possible over spectrogram based algorithms. Simulations and preliminary detection performance results (ROC's) are presented using a mathematical model from the literature of the North Atlantic Right Whale (NARW), along with numerical acoustic propagation software.

3aUWb17. Broadband class I flextensional transducer. Yu Lan, Wei Lu, Yongjie Sang, and Tianfang Zhou (Harbin Eng. Univ., Nantong St. No.145, Hei Longjiang Province, Harbin, Harbin 0086, China, lanyu_2013@126.com)

Researches about high-power, low-frequency projectors are the most important technology for long range sonar systems. Flextensional transducer is a typical low-frequency, high-power projector, which has the advantages of small volume and light weight due to displacement amplification. In this paper, attention is focused on expanding bandwidth of the flextensional transducer. Class I flextensional transducer usually consists of a slotted convex shell and a driving stack. There are two kinds of main vibration modes at low frequency, the first flexural mode and the membrane mode. They can be coupled and broadband transmitting response was obtained. Finite element (FE) techniques are applied to the design of this multiple-mode-coupling transducer. Finite element model was made with ANSYS software and it can be used to do vibration mode analyze, operative mode simulation and structure optimization. Base on the analyze results, the multiple-mode-coupling flextensional transducer was designed and tested. Key words: flextensional transducer; broadband; multiple-mode-coupling; Finite Element Method

3aUWb18. Doubly resonant underwater acoustic transducer for long distance sound propagation. Andrey K. Morozov (Teledyne, 49 Edgerton Dr., North Falmouth, MA 02556, amorozov@teledyne.com)

The presentation describes the design and test of innovative deep water low frequency sound source for long range acoustic communications and navigation. The light-weight, low frequency (200–1000 Hz), broadband underwater sound source comprises an inner resonator tube with thin walls tuned to a certain frequency surrounded by a shorter, larger-diameter, lower frequency tuned outer resonator tube. These resonating tubes are opened on both ends and made of carbon fiber. The tubes are asymmetrically shifted along the main axis and sound pressure can penetrate from internal pipe though the area under the shifted external pipe into that external pipe and back. By changing length of shifted area the coupling coefficient of two resonators can be regulated to achieve a necessary bandwidth. The design uses depth independent oil-filled acoustic transducer. The transducer is light, operational for all ocean depth, and reasonably broadband. A multiresonant systems usually need a precise complicated adjustment of their parameters to get necessary bandwidth with limited variability of frequency response inside the frequency range. It can be even more complicated, when design includes new, never tested before, materials. Application of finite element analysis allowed to predict necessary parameters and avoid a long series of water tests with parameters adjustment.

3aUWb19. Optimizing eigenray generation for real time simulations. Andrew J. Fredricks (Undersea Warfare Weapons, Vehicles, and Defensive Systems, Naval Undersea Warfare Ctr., 487 NewBedford Rd., Rochester, MA 02770, fredra@comcast.net) and Sheri Martinelli (Undersea Warfare Weapons, Vehicles, and Defensive Systems, Naval Undersea Warfare Ctr., Newport, RI)

Real time simulation is a critical supplement to in-water testing for the assessment of naval system performance. Limitations exist on the number of independent runs, on geographical locations in which in-water tests can be performed, and on the amount of control experimenters possess over the test environment. Many systems have critical time-sensitive functionality (e.g., acoustic homing) which constrains the ability to produce realistic time series for injection; but a reduced fidelity solution can still be of use. Graphics hardware (GPU) has become a significant computing platform in its own right. Its application requires a mapping of the propagation algorithm to the GPU computing paradigm and careful tweaking to squeeze out maximum performance. We will look at taking from theory just what we need to hand-tune code for a GPU + CPU computing platform, and the limitations of a high speed, range dependent, eigenray code. We also consider a related approach that uses the resulting eigenrays to initialize an iterative method which updates the eigenray solution as the source and receiver update their relative positions. [Work funded by the Office of the Secretary of Defense, Test Resource Management Center's Resource Enhancement Project element of the Central Test and Evaluation Investment Program.]

3aUWb20. Acoustic propagation in surface channel formed by low salinity water in the East China Sea and the tropical Atlantic Ocean. Juho Kim, Hansoo Kim, Dong-guk Paeng (Dept. of Ocean System Eng., Jeju Natl. Univ., Ara 1 dong, Jeju 064-756, South Korea, lizard@jejunu.ac.kr), and Jongkil Lee (Mech. Eng. Education Dept., Andong Natl. Univ., Andong, South Korea)

Salinity is usually neglected in underwater acoustics because of its minimal contribution on sound speed variation. However, it is required to consider in calculation of sound speed and acoustic propagation for low salinity water freshened by continental run-off. Furthermore, the acoustic characteristics of low salinity environment are not fully investigated yet. Therefore, regional difference of acoustic propagation in low salinity water was studied by comparing the acoustic characteristics of the East China Sea with those of the tropical Atlantic Ocean in this paper. Frequency dependency of sound propagation in the haline channel was analyzed with transmission losses and low frequency cut-off using oceanic data from NODC (National Oceanographic Data Center). The tropical Atlantic Ocean showed larger channel depth, critical angles and less transmission loss in the haline channel than the East China Sea. The cut-off frequency in the haline channel was computed as around 1 kHz and 5 kHz in the tropical Atlantic Ocean and the East China Sea, respectively. The effects of low salinity water on sound propagation showed regional characteristics, and need to be considered in sonar operation near sea surface. [This work was supported by Defense Acquisition Program Administration and Agency for Defense Development under the contract UD130007DD.]

3aUWb21. Simulation of underwater environments using the Discrete Huygens Modeling. Renato T. de Carvalho, Gregário G. Azevedo, and Júlio A. Cordioli (Mech. Eng., Universidade Federal de Santa Catarina, Campus Universitário Reitor João David Ferreira Lima, Florianápolis 88040-900, Brazil, renato@lva.ufsc.br)

The Discrete Huygens Modeling (DHM) is a numerical method that applies the Huygens principle to a discretized medium in order to provide time domain solutions to wave propagation problems. The method was originally conceived and applied in Electromagnetism, but previous works have shown that DHM is also a viable and promising approach for the simulation of acoustic problems. The main advantages of the method are its reduced computation cost when compared with other numerical methods and its relatively easy implementation and parallelization. However, little can be found in the literature about the use of DHM to underwater acoustic problems, and previous work have been limited to acoustic propagation only in the water column. In this work, a simple model of the acoustic field generated in the water column by a monopole was created, and the prediction of transmission, reflection, and scattering processes at the interface with the sediment were carried out. A DHM code has been fully implemented allowing the modeling of multiple media. The results obtained with the DHM model were compared with a Finite Element model yielding very good agreement, while the DHM approach proved to be much faster.

3aUWb22. An error reduction method for double-plane nearfield acoustic holography based on boundary element method. Dejiang Shang and Yongwei Liu (College of Underwater Acoust. Eng., Harbin Eng. Univ., Rm. 1205, SHUI SHENG Bldg., 145 Nantong St., Nangang District, Harbin 150001, China, shangdejiang@hrbeu.edu.cn)

Nearfield planar acoustic holography method based on BEM is often used in experiments because it is very convenient to scan the sound field on planes. However, the sound field reconstruction error on the sound source surface is usually very big due to the dimension limitation of measuring planes. The method proposed here, by measuring some extra far field points out of the planes, which are combined with those on the planes to reconstruct the sound field on the source surface, can reduce this sort of error. The numerical simulation has been carried out by this method for a spherical sound source with radius of 1.0 m. The double planes are located at $x=\pm 0.6$ m, assumed that the sound source is at (0,0,0) in Cartesian coordinate. The max reconstructing error of sound pressure on the sound source surface can be reduced from 6 dB to less than 2 dB at frequency 500 Hz when we add two extra far field points at ($0,\pm 10$ m,0). Obviously, this method is very meaningful in practical experimental tests. More numerical simulation by

this method for more complex sound source field reconstruction will be shown in the full paper.

3aUWb23. An equivalent physical model of three acoustic waves interaction in nonlinear medium. Desen Yang (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, China), Haoyang Zhang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No.145 Nantong St., Harbin 150001, China, zhanghaoyang@hrbeu.edu.cn), Shengguo Shi (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, China), Wei Jiang (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China), and Jie Shi (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, China)

Among the parametric effects, three waves interaction has been widely investigated in many fields of physics, such as nonlinear optics and plasma. The interaction of three acoustic waves in the medium with quadratic nonlinearity, which must satisfy conditions on their frequencies and wavenumbers, is essentially confined to energy exchange between three components. Based on Burgers equation, the propagation rules of three acoustic waves, which are called a resonant triad, are governed by a drastically simplified system of three coupled nonlinear ordinary differential equations. A mathematical equivalence between the equations for an acoustic triad and a simple parametric vibration system, the undamped elastic pendulum, is discussed in this paper by a multiple time-scale analysis. We study the dynamics of this system, drawing analogies between its behavior and that of the acoustic triad. Finally, it is certified that three acoustic waves interaction can be described by Mathieu type equations in case one acoustic wave is much stronger than the others in three waves. This means that experiments with an elastic pendulum can give us new insights into dynamics of mechanism in three acoustic waves interaction.

3aUWb24. Theory and experiment of the resonance sound wave interactions in water. Desen Yang (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, China), Wei Jiang (College of Underwater Acoust. Eng., Harbin Eng. Univ., No. 145, Nantong St., Heilongjiang, Harbin HL 451, China, jiangwei_first@126.com), Shenguo Shi (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Harbin, China), Haoyang Zhang, and Shiyuan Jin (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin, China)

Equations governing nonlinear interactions among acoustic waves in water which satisfy the resonance conditions are derived from the Burgers equation. Taking into account of the second-order nonlinear, the three-wave interaction is the fundamental process of the interaction equations. Then, taking the three-wave interaction equation as an example, the energy transfer mechanism among the three acoustic waves is quantitatively analyzed. And the three-wave sound energy propagation is studied through numerical calculation. An interesting phenomenon is found that, when the three acoustic waves meet the relationship of a weak low-frequency sound and two strong high-frequency sounds, the energy of low-frequency sound will be amplified or reduced in some regions during the three-wave propagation. And the location and size of the regions are affected by the acoustic wave amplitude, frequency, and phase. The variation severities of low-frequency acoustic wave energy are mainly determined by other two high-frequency acoustic wave energy. The frequencies of two high-frequency sound waves have little effects on the energy of low-frequency sound wave. The region whether is amplification or reduction is determined by the phase difference of three waves. The variation laws of low-frequency sound energy are also verified by the experimental results in river.

3aUWb25. Predicting range-frequency interference patterns from broadband sources in a range-dependent, continental shelf environment. Alexander W. Sell (Acoust., Penn State Univ., 830 Cricklewood Dr., Apt. 207, State College, PA 16803, aws164@psu.edu) and R. Lee Culver (Acoust., Penn State Univ., University Park, Oregon)

The 2007 CALOPS dataset contains horizontal line array received signals from numerous surface vessels transiting an area off the coast of southeast Florida. Parabolic equation propagation models using measured bathymetric, sound velocity profile, and sediment parameter data suggest that substantial mode coupling should be occurring for several down-slope propagation scenarios in the dataset. However, received acoustical data for these scenarios, in the form of range-frequency interference patterns, are best described by adiabatic propagation. This discrepancy has implications that affect waveguide invariant parameter estimation, which is necessary for invariant-based passive ranging. The causes of the incongruous model output will be discussed, as well as their effects on estimating waveguide invariant distributions for these scenarios. [This research was supported by the Applied Research Laboratory, at the Pennsylvania State University through the Eric Walker Graduate Assistantship Program.]

3aUWb26. Coherence measurements of acoustic normal modes during one month of internal wave events on the New Jersey continental shelf. Lin Wan and Badiey Mohsen (College of Earth, Ocean, and Environment, Univ. of Delaware, 003 Robinson Hall, Newark, DE 19716, wan@udel.edu)

During the Shallow Water Acoustic Experiment 2006 (SW06) conducted on the New Jersey continental shelf, the three-dimensional (3D) temperature field for one month of internal wave (IW) events has been reconstructed by Badiey *et al.* [J. Acoust. Soc. Am. El. **134**(5), 4035 (2013)]. The aforementioned IW events, with the angle between the acoustic track and the IW front varying from -8° to 83°, were measured while simultaneously acoustic signals were transmitted from fixed sources at an along-shelf distance of about 20 km with frequencies at 87.5–112.5 Hz (m-sequence), 175–225 Hz (m-sequence), 270–330 Hz (chirp), and 470–530 Hz (chirp), respectively. The acoustic signals were recorded by an L-shaped hydrophone array moored inside the area with IW measurements. The main goal of this paper is to analyze the coherence of acoustic normal modes accompany with the simultaneously measured IW events. The coherence of acoustic normal modes decomposed from the measured acoustic field is obtained as a function of frequency and IW parameters, such as the IW propagation direction, amplitude, coherence length, etc. The relationship between the modal coherence and IW parameters is discussed. [Work supported by ONR322OA.]

3aUWb27. Fluctuations of arrival time and frequency shifts of a wideband sound signal in the presence of coastal internal Kelvin waves in shallow water. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru) and Andrey Lunkov (Wave Res. Ctr., Geneal Phys. Inst., Moscow, Russian Federation)

Internal Kelvin waves (IKWs) in a near shore area of shallow water with a curvilinear coastal line (gulf, lake, and channel) where the radius of curvature can be about 5–30 km are considered as a reason for the sound field variability. Amplitude of the thermocline vertical displacement due to IKW in such an area changes along the distance from the center of this circle (minimal value, about null) to the coast (maximal value, up to 5 m). Numerical simulations are implemented to study the effect of IKWs on the propagation of acoustic waves generated by a wideband (0.7 to 1 kHz) sound source. The results of the simulations show that the signal propagation time changes up to 1% following the thermocline displacement caused by IKW. Also, considerable shifts (more than 30% of the central frequency) of the interference pattern in the frequency domain are observed. The shifts change with distance from the shore and have opposite directions for positions of a single hydrophone receiver above or below the thermocline. Analytical estimates are obtained; experimental setup and a possible scheme of monitoring IKW are discussed. [Work was supported by RFBR and BSF.]

3aUWb28. Influence of short time scale water column fluctuations on broadband signal intensity. Justin Eickmeier and Badiey Mohsen (CEOE, Univ. of Delaware, 17 McCord Dr., Newark, DE 19713, jeickmei@udel.edu)

The KAM11 experiment was conducted in 100 m of water off the Western side of Kauai, Hawaii, in June/July, 2011 during which two identical bottom mounted tripods, separated by 1 km, transmitted reciprocal chirp sequences. A monitoring hydrophone was suspended at a depth of 25 m from the *R/V Kilo Moana* approximately midway between the tripod stations. Impulse response measurements at the suspended hydrophone show intermediate arrivals between direct-path/bottom-bounce and surface-bounce arrivals unique to only one tripod source. Focusing and defocussing of the intermediate arrivals show the influence of fluctuations in the sound speed profile over time scales from seconds to hours. Short term intensity variations arise from surface driven vertical undulations in the water column. Over longer time scales changes in the thermocline govern the evolution of the intermediate arrival. Data model comparison is conducted with ray tracing and parabolic equation modeling. [Work supported by ONR 322OA.]

3aUWb29. Model and data comparisons of ocean acoustic intensity statistics in the Philippine Sea 2010 experiment. Andrew A. Ganse, Rex K. Andrew, Frank S. Henyey, James A. Mercer (Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, aganse@apl.washington.edu), Peter F. Worcester, and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California, San Diego, La Jolla, CA)

The statistics of intensity fluctuations in dual-band, low-frequency, ocean acoustic transmissions across 500 km in the Philippine Sea 2010 experiment appear to be at odds with the standard micro-multipath theory predicting them. M-sequence encoded acoustic signals at 200 Hz and 300 Hz were transmitted for 54 h from a ship-suspended multiport source to a distributed vertical line array (DVLA) with 149 working hydrophones covering most of the 5500 m water column. Histograms of the received intensity fluctuations are approximately exponential. Intensity fluctuations at the two frequencies are strongly correlated. Scintillation indices and variances of log intensity are high (>1 and >5.57dB, respectively), suggesting the measurements are not yet in a saturated regime but still significantly scattered. However, for all this, the pulses received at the DVLA are virtually the same width as those recorded next to the source; no pulse spreading or arrival splitting is seen. So, the fading observed in the data does not appear to match this causal mechanism. To explore the connection with micro-multipath theory more fully, we compare the intensity statistics from the Philippine Sea 2010 data with those calculated both from simulated random oceans and from Flatté and Rovner's path integral theory. [Work supported by ONR.]

3aUWb30. Stochastic characterization of acoustic signals for sequential dispersion tracking and geoacoustic inversion. Nattapol Aunsri and Zoi-Heleni Michalopoulou (Mathematical Sci., New Jersey Inst. of Technol., 323 ML King Blvd., Newark, NJ 07102, michalop@njit.edu)

In previous work, we had shown how sequential Bayesian filtering methods can be used for the successful extraction of dispersion curves from broadband long-range acoustic data. Here, we extend this work by tracing carefully the true nature of the noise and the resulting probability density observations of the spectrogram of the received time series, employed in the tracking. The Gaussian model typically used in instantaneous frequency tracking relies on the assumption that noise is additive in the frequency domain. This model is, however, inaccurate. We discuss a chi-squared model of the acoustic data perturbations and its role in dispersion curve tracking. The new method provides much clearer curves than those computed with previous approaches. We demonstrate the potential of the technique by applying it to synthetic and real data for dispersion curve estimation and bathymetry and sediment sound speed inversion. [Work supported by ONR.]

3aUWb31. Low frequency sound absorption in the Arctic Ocean: Potential impact of ocean acidification. David Browning (139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com), Peter D. Herstein (24 Mohegan Rd., Charlestown, RI), Peter M. Scheifele (Dept. of Comm. Sci., Univ. of Cincinnati, Cincinnati, OH), and Raymond W. Hasse (145 Old Colchester Rd., Quaker Hill, CT)

The principal mechanism for low frequency absorption in seawater is a boron reaction that is *p*H dependent; the lower the *p*H, the lower the absorption. Twenty seven years ago, Mellen *et al*. [J. Acoust. Soc. Am. **82**, S30 (1987)] computed the low frequency sound absorption for the Arctic Ocean. Since the time the carbon dioxide (CO_2) level in the atmosphere has been continually increasing. Experts predict that the resulting ocean acidification may increase by up to 170% this century. Acoustically, the Arctic Ocean is most sensitive to rapid change not only because the cold water readily absorbs CO_2, but also because the sound channel axis is at or near the surface. The range of reduction in the low frequency sound absorption is presented based on possible future acidification scenarios, mindful that this is just one component of a complex evolution that is occurring in the Arctic.

3aUWb32. Sensing stratified turbulence and shear instability using an acoustic high frequency broadband backscatter array. Jonathan R. Fincke (AOSE, MIT/WHOI Joint Program, 99 Hancock St., Apt. 10, Cambridge, MA 01239, jfincke@mit.edu) and Andone Lavery (AOPE, Woods Hole Oceanographic Inst., Woods Hole, MA)

High-frequency broadband acoustical backscattering measurements with an array of transducers have allowed the temporal and spatial evolution of shear instabilities in a strongly stratified estuarine environment to be investigated. Development of accurate remote sensing techniques for estimating mixing are of significant interest to the geophysical fluid dynamics community. An array of six high-frequency (120–600 kHz) broadband and narrow beam width (1 to 6 degrees half-beamwidth, depending on the frequency) transducers spaced 1.2 m apart were deployed in the Connecticut River estuary both in the along stream and across stream direction to observe high Reynolds number stratified shear instabilities. In this presentation, results of these high-resolution temporal and spatial sampling measurements of shear instabilities are shown. These measurements demonstrate the utility of these techniques for improving our understanding of the evolution of shear instabilities.

3aUWb33. Is low frequency sound level uniformly increasing on a global scale? Jennifer L. Miksis-Olds (Appl. Res. Lab, Penn State, PO Box 30, Mailstop 3510D, State College, PA 16804, jlm91@psu.edu)

Deep water ambient sound level increases have been documented in the eastern North Pacific Ocean over the past 60 years. It remains unclear whether this increasing trend is observed in other regions of the world. In this work, data from the Comprehensive Nuclear Test Ban Treaty Organization International Monitoring System (CTBTO IMS) were used to examine the rate and direction of low frequency sound level change over the past decade in the Indian, South Atlantic, and Equatorial Pacific Oceans. The sources contributing to the overall sound level patterns differed between the regions. The dominant source observed in the South Atlantic was sound from seismic air gun surveys, while shipping and biologic sources contributed more to the acoustic environment at the Equatorial Pacific location. Unlike the increasing trend observed in the NE Pacific, sound levels over the past 5–6 years in the Equatorial Pacific were decreasing. Decreases were also observed for specific sound level parameters and frequency bands in the South Atlantic Ocean. Based on these observations, it does not appear that low frequency sound levels are increasing in all regions of the worlds' oceans. [Work supported by the Office of Naval Research.]

3aUWb34. Passive acoustics embedded on gliders—Weather observation through ambient noise. Pierre Cauchy, Pierre Testor, Laurent Mortier (LOCEAN, 4 Pl. jussieu, Paris 75252, France, pierre.cauchy@gmail.com), Laurent Beguery (DT INSU, Toulon, France), and Marie-Noelle Bouin (CMM, Brest, France)

Underwater gliders can provide high resolution water temperature and salinity profiles. Being able to associate them with a surface weather conditions estimation would allow to better study sea-air interactions. Since *in-situ* observations of the marine meteorological parameters are difficult, the development of a glider embedded weather sensor has been studied, based on the WOTAN approach. In the 1–30 kHz frequency range, the background underwater noise is dominated by wind generated noise. Focusing on the sound pressure level at 5, 8, 10, and 20 kHz allows to estimate the wind speed. Thus, deploying a glider with an embedded hydrophone gives an access to the surface weather conditions around its position. We have deployed gliders in the Mediterranean sea, with passive acoustic monitoring devices onboard. Four months of data have been recorded. Wind speed estimations have been confronted to weather buoys observations and atmospheric models predictions. Wind estimates have been obtained with a ~2 m/s

error. A specific emphasis has been placed on the robustness of the processing through multi frequencies analysis and depth induced attenuation correction. A downscaling study has been performed on the acoustic sampling protocol, in order to meet the low energy consumption glider standards, for a future real time embedded processing. The glider generated noise and its vertical movement are not perturbing the estimation. Moreover, the surface behavior of the Slocum gliders allows an estimation of the wind direction.

3aUWb35. Hydroacoustic signals of Antarctic origin detected at ocean-bottom seismic stations off New Zealand. Justin S. Ball and Anne F. Sheehan (Geological Sci., CIRES/ Univ. of Colorado, 2200 Colorado Ave. #399, Boulder, CO 80309, justin.ball@colorado.edu)

Glacial calving from polar ice sheets is an important indicator of global climate change, and knowledge of ice discharge rates is useful for predicting global sea level variability and deep ocean circulation patterns. Since calving events are difficult to observe *in-situ* due to the remoteness of polar regions, the remote detection of hydroacoustic signals originating from these events is a useful monitoring tool for climatologists. We have observed hydroacoustic T-phases on an Ocean Bottom Seismic (OBS) network of 30 seismometers and differential pressure gauges that was deployed off the South Island of New Zealand in 2009–2010. These signals are emergent and strongly dispersive, containing most of their energy in the band between 5 and 15 Hz. We estimated backazimuths for events recorded on differential pressure gauges using array beamforming and F-K analysis. Our results suggest that some observed events originate in the vicinity of Ninnis and Mertz glaciers on George V Coast, East Antarctica, in general agreement with epicenters located by prior surface-wave based calving detection studies.

3aUWb36. Speculations on the cause of finite density and sound speed gradients near the interface of muddy sediments. Allan D. Pierce (Retired, PO Box 339, 399 Quaker Meeting House Rd., East Sandwich, MA 02537, allanpierce@verizon.net), William L. Siegmann (Mathematical Sci., Rensselaer Polytechnic Inst., Troy, NY), and Joseph O. Fayton (Mathematical Sci., Rensselaer Polytechnic Inst., Portsmouth, Rhode Island)

Holland et al. [JASA (2013)] at the previous ASA meeting reported positive density gradients and negative sound speed gradients at the water-mud sediment interface and asked "what processes drive them." A derivation using the Mallock-Wood relations yields a simple high-porosity approximate relation giving a negative proportionality between sound speed gradient and density gradient independent of depth and porosity. It is also argued that the solid portion of the mud consists of tiny mineral platelets, each of which carries a net negative charge. The presence of dissolved salts causes each platelet to behave as an electrical quadrupole, so that there is, on the average, a small electrical repulsive force between the platelets. With gravity taken into account, the equilibrium separation distance between two parallel vertically aligned platelets is one to two orders of magnitude greater than a typical length scale of the face of a platelet. However, when platelets touch, edge to face, there is an attractive force between platelets, and the net effect is that the platelets tend to be separated at a much shorter distance than the stand-off distance deep within the sediment. Paper reports ongoing efforts to estimate the depth over which the transition occurs.

WEDNESDAY AFTERNOON, 7 MAY 2014 — 555 A/B, 1:30 P.M. TO 3:05 P.M.

Session 3pAAa

Architectural Acoustics: The Technical Committee on Architectural Acoustics Vern O. Knudsen Distinguished Lecture

David Lubman, Chair
dlacoustics, 14301 Middletown Ln., Westminster, CA 92683

Chair's Introduction—1:30

Invited Paper

1:35

3pAAa1. Is there any acoustical reason that supports non-rectangular concert hall design? Tapio Lokki (Media Technol., Aalto Univ., POBox 15500, Aalto 00076, Finland, Tapio.Lokki@aalto.fi)

The debate between shoe-box and non-shoe-box concert halls has been around for several decades. From the total concert experience point-of-view, there seems to be pros and cons for both designs. However, when only the acoustics of a hall are considered, the shoe-box design is very often preferred. This presentation discusses the basic differences between these two hall types, and how they affect, e.g., perceived bass, envelopment, clarity, intimacy, and loudness. The presentation is supported by recent data that was gathered in a large listening test during winter 2014. Typical sensory profiles of these two hall types are presented with links to listening test subjects' preferences. Moreover, the presentation will explain why shoe-box halls render larger dynamic range than other halls do. Even though room acoustics, defined with an impulse response, is linear, non-linear dynamic differences exist between halls due to the non-linear excitation (an orchestra) and non-linear human spatial hearing.

Session 3pAAb

Listening to the "Virtual Paul's Cross"—Auralizing 17th Century London II

Matthew Azevedo, Chair
Acentech Inc., 33 Moulton St., Cambridge, MA 02138

The purpose of this session is to provide an opportunity for people to listen to the Virtual Paul's Cross auralization, which allows listeners to experience John Donne's 1622 Gunpowder Day sermon while surrounded in three dimensions by a reactive crowd of up to five thousand, the bells of St. Paul's, and the ambient soundscape of 17th century London. The auralization allows for real-time changes to crowd size, listener position, the behavior of the intelligent agents which create the crowd reactions, and variations in the type and frequency of ambient sounds and requires over one hundred concurrent audio channels, a dozen channels of real-time convolution, hours of project-specific source recordings, and a complex network of intelligent and stochastic logical structures.

Session 3pAO

Acoustical Oceanography: Acoustical Oceanography Prize Lecture

Timothy K. Stanton, Chair
Woods Hole Oceanogr. Inst., MS #11, Woods Hole, MA 02543-1053

Chair's Introduction—1:00

Invited Paper

1:05

3pAO1. Advances in remote inference of physical and biological parameters using acoustic scattering techniques: Mapping the ocean in broadband "color". Andone C. Lavery (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 98 Water St., MS 11, Bigelow 211, Woods Hole, MA 02536, alavery@whoi.edu)

Active narrowband acoustic scattering techniques have been used for decades to infer the distribution of marine organisms, such as fish and zooplankton, and to image physical processes, such as bubbles, suspended sediments, internal waves, and microstructure. Accurately inferring relevant biological and physical parameters from the acoustic returns, such as size or abundance of organisms or intensity of mixing, has represented a far more formidable obstacle, requiring a multi-faceted approach in order to make significant headway. Over the years, advances have been made in understanding the fundamental scattering physics, resulting in more robust and accurate scattering models. These models have been guided and tested by controlled laboratory scattering experiments as well as in a plethora of field experiments. Rapid advances in instrumentation and deployment platforms have also enabled new insights to be gained. In this presentation, a brief overview of this research area is given, results from the development and implementation of broadband scattering techniques for studying physical and biological processes over relevant spatial and temporal scales are presented, and limitations of these techniques considered. Possible future directions and advances in the area of remote physical and biological parameter estimation from active acoustic scattering data will be discussed.

Session 3pBA

Biomedical Acoustics: Nonlinear Response of Encapsulated Microbubbles

Kausik Sarkar, Chair
Mech. and Aerosp. Eng., George Washington Univ., 801 22nd St. NW, Washington, DC 20052

Contributed Papers

1:00

3pBA1. Effects of ambient pressure variation on the subharmonic response from contrast microbubbles. NIma Mobadersany (George Washington Univ., Washington, Virginia), Amit Katiyar (Mech. Eng., Univ. of Delaware, Austin, Texas), and Kausik Sarkar (George Washington Univ., 801 22nd St. NW, Washington, DC 20052, sarkar@gwu.edu)

Ambient pressure dependent subharmonic response scattered from encapsulated contrast agent is investigated for non-invasive estimation of local blood pressure. Bubble dynamics is simulated using two different models of the encapsulation: the strain-softening elastic model and the Marmottant model. Unlike fundamental response, subharmonic response from a bubble occurs in a narrow range of excitation pressures—it appears only above a threshold excitation and disappears at higher excitation. The variation in subharmonic responses is chaotic with increasing ambient pressure at very low excitation frequencies. On the other hand, for high excitation frequencies, the subharmonic response increases with increasing ambient pressure For intermediate frequencies, the variation can be either monotonic increase, monotonic decrease, or nonmonotonic with increasing ambient pressure depending upon the excitation intensity. The simulated results will be discussed taking into account the effects of ambient pressure variation on the subharmonic threshold. [Partially supported by National Science Foundation.]

1:15

3pBA2. Ambient pressure estimation using subharmonic emissions from contrast microbubbles. Krishna N. Kumar, Shirshendu Paul (George Washington Univ., 801 22nd St. NW, Washington, VA 20052, krishnagwu@gwu.edu), and Kausik Sarkar (George Washington Univ., Washington, DC)

In cancerous tumors, the interstitial fluid pressure is higher than that in normal tissues, and therefore can be used as a diagnostic marker. Here we are presenting the results of an *in vitro* study aimed at developing an ultrasound-aided noninvasive pressure estimation technique using contrast agents—Definity®, a lipid coated microbubble, and an experimental poly lactic acid (PLA) microbubbles. Scattered responses from these bubbles have been measured *in vitro* as a function of ambient pressure using a 3.5 MHz acoustic excitation of varying amplitude. Definity bubbles produced stronger subharmonic than the PLA coated ones, and therefore, are better suited for this application. At an acoustic pressure of 500 kPa, Definity® microbubbles showed a linear decrease in subharmonic signal with increasing ambient pressure, registering a 12 dB reduction at an overpressure of 120 mm Hg. However, at other frequencies, the variation of subharmonic emission with ambient pressure is nonmonotomic as was also predicted by theoretical modeling in our group. [Partially supported by National Science Foundation.]

1:30

3pBA3. Acoustic characterization of polymer-encapsulated microbubbles with different shell-thickness-to-radius ratios using *in vitro* attenuation and scattering: Comparison between different rheological models. Lang Xia (Mech. and Aerosp. Eng., George Washington Univ., Washington, VA), Shirshendu Paul (Mech. and Aerosp. Eng., George Washington Univ., Washington, DC), Parag V. Chitnis, Jeffrey Ketterling (Lizzi Ctr. for Biomedical Eng., Riverside Res., New York, NY), Paul Sheeran, Paul Dayton (Biomedical Eng., Univ. of North Carolina Chapel Hill, Chapel Hill, NC), and Kausik Sarkar (Mech. and Aerosp. Eng., George Washington Univ., 801 22nd St. NW, Washington, DC 20052, sarkar@gwu.edu)

Acoustic behaviors of five different polymer (polylactide) encapsulated microbubbles—PB127 (Point Biomedical), PH37, PH43, PH44, PH45 (Phillips Healthcare) with different shell-thickness-to-radius ratios (STRR) of 3.5, 30, 40, 65, and 100 nm/μm have been characterized. *In vitro* attenuation data were used to determine the interfacial rheological properties of their shells. Use of different models—Newtonian, viscoelastic, strain-softening, Marmottant, and Church—resulted in similar rheological properties. The shell elasticity and shell viscosity were found to increase with increasing shell thickness as expected. The nonlinear scattered response from these microbubbles were measured. Experimentally measured scattered subharmonic response were compared with the model predictions.

1:45

3pBA4. Nonlinear intravascular ultrasound contrast imaging with a modified clinical system. Himanshu Shekhar, Ivy Awuor, Sahar Hashemgeloogerdi, and Marvin M. Doyley (Elec. and Comput. Eng., Univ. of Rochester, 212 Conant Rd. Apt. C, Rochester, NY 14623, himanshushekhar@rochester.edu)

An intravascular ultrasound system capable of visualizing microbubble contrast agents could provide functional information for assessing atherosclerotic plaques. The goal of this study was to investigate the feasibility of contrast-enhanced imaging with a modified commercial intravascular ultrasound system. We employed an iLab^{TM} system (Boston Scientific/Scimed, Natick, MA) equipped with an Atlantis^{TM} PV imaging catheter (15-MHz center frequency, 26% fractional bandwidth) to image tissue mimicking phantoms that had contrast agent (Targestar-P®, Targeson Inc., CA) flowing in side channels parallel to the center lumen. Chirp-coded pulses were employed with transmit frequency of 12 MHz and peak pressures ranging from 1–2 MPa. The ultraharmonic response (18 MHz) was isolated from the backscattered radio-frequency using pulse inversion and matched filtering, to produce contrast specific images. We evaluated the detection sensitivity of the agent as a function of microbubble concentration and transmit pulse parameters. The results revealed that side channels with diameters ranging from 500 μm to 2 mm could be visualized for a wide range of concentrations. These results demonstrate that functional imaging of plaque neovascularization is feasible with commercially available intravascular catheters. Further development of such systems can facilitate the widespread use of contrast-enhanced intravascular ultrasound for preclinical research and clinical imaging.

2:00

3pBA5. Microbubble spectroscopy of microbubble-loaded stem cells for targeted cell therapy. Tom Kokhuis, Ilya Skachkov (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Benno Naaijkens (Dept. of Pathol., VU Univ. MC, Amsterdam, Netherlands), Lynda Juffermans (Dept. of Physiol., VU Univ. MC, Amsterdam, Netherlands), Otto Kamp (Dept. of Cardiology, VU Univ. MC, Amsterdam, Netherlands), Ton van der Steen (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands), Michel Versluis (Phys. of Fluids group, Univ. of Twente, P.O. Box 217, Enschede 7500 AE, Netherlands, m.versluis@utwente.nl), and Nico de Jong (Biomedical Eng., Erasmus MC, Rotterdam, Netherlands)

Stem cells can be conjugated with targeted microbubbles to form highly echogenic complexes, dubbed StemBells. The complexes can improve stem cell delivery for the local repair of damaged cardiac tissue after a myocardial infarction through propulsion by acoustic radiation forces. While the first *in-vivo* tests hold great promise, the system would greatly benefit from a mapping of the acoustic parameter space. Here, we develop the theoretical background based on a modified Rayleigh-Plesset type equation to describe the dynamics of the StemBells in response to ultrasound. The complex is shown to resonate as a whole entity and resonance curves are constructed from numerical simulations resembling single bubble responses at a size that relates to the effective complex radius ~10 μm. Ultra high-speed optical imaging of single StemBell complexes at different frequencies using the microbubble spectroscopy method allows for a full characterization with excellent agreement with the developed model. Moreover, from the experimental resonance curves, we obtain values for the effective viscoelastic shell parameters of the StemBell complexes. These results have enabled the demonstration of the feasibility of manipulating StemBells inside chicken embryo microvasculature in an accompanying paper.

2:15

3pBA6. StemBells: Localized stem cell delivery using targeted microbubbles and acoustic radiation force. Tom Kokhuis, Ilya Skachkov (Biomedical Eng., Erasmus Medical Ctr., ’s-Gravendijkwal 230, Faculty Bldg. (Rm. Ee2302), Rotterdam 3000 CA, Netherlands, t.kokhuis@erasmusmc.nl), Benno Naaijkens (Dept. of Pathol., VU Univ. Medical Ctr., Amsterdam, Netherlands), Lynda Juffermans (Dept. of Physiol., VU Univ. Medical Ctr., Amsterdam, Netherlands), Otto Kamp (Dept. of Cardiology, VU Univ. Medical Ctr., Amsterdam, Netherlands), Antonius van der Steen (Biomedical Eng., Erasmus Medical Ctr., Rotterdam, Netherlands), Michel Versluis (Phys. of Fluids Group, Univ. of Twente, Enschede, Netherlands), and Nico de Jong (Biomedical Eng., Erasmus Medical Ctr., Rotterdam, Netherlands)

The use of stem cells for regenerative tissue repair is hampered by the low number of cells delivered to the site of injury. To increase the delivery, we developed a new technique in which stem cells are coated with functionalized microbubbles, creating echogenic complexes dubbed StemBells. StemBells are highly susceptible to acoustic radiation force; this acoustic force can then be used after injection to deliver the StemBells locally at the treatment site. The dynamics of StemBells during ultrasound insonification was characterized using high-speed optical imaging and is described in an accompanying paper. Here, we investigate the feasibility of manipulating StemBells acoustically after injection employing a chicken embryo model, allowing for the real-time optical observation of the effects of acoustic radiation force *in vivo*. StemBells were infused by placing a custom-made catheter into one of the vitelline veins. Acoustic radiation force (1 MHz, P = 200–450 kPa, 10% duty cycle) was observed to propel StemBells from the centerline of the microvessels (200–500 μm) to the wall distal from the transducer. Peak translational velocities increased with pressure and varied between 50 μm/s to 300 μm/s. The acoustic radiation force had no effect on the trajectory of bare stem cells.

2:30

3pBA7. Frequency-sum passive cavitation imaging. Kevin J. Haworth, Kirthi Radhakrishnan (Internal Medicine, Univ. of Cincinnati, Biomedical Ultrason. and Cavitation Lab., Cincinnati, OH, kevin.haworth@uc.edu), and T. Douglas Mast (Biomedical Eng., Univ. of Cincinnati, Cincinnati, OH)

Passive cavitation imaging (PCI) is a method for spatially mapping acoustic emissions caused by microbubble activity, including subharmonic and ultraharmonic emissions that denote stable cavitation. The point spread function (PSF) of passive cavitation images is diffraction limited. When typical clinical diagnostic linear arrays are used for PCI, the diffraction limit results in high azimuthal resolution but low axial resolution. Abadi *et al.* (2013)recently demonstrated a method called frequency-sum beamforming, which employs second-order or higher products of the acoustic emissions to manufacture higher frequencies, thereby reducing the size of the PSF. We applied this approach to cavitation emissions recorded from albumin-shelled bubbles insonified by 2 MHz ultrasound. Cavitation emissions were recorded on a 5 MHz, 128 element linear array using a Vantage scanner (Verasonics Inc.). Quadratic and fourth-order frequency-sum beamforming was applied to both harmonic and ultraharmonic cavitation emissions. Corresponding simulations were also performed to illustrate frequency-sum passive cavitation imaging of multiple bubbles. In comparison to delay-and-sum PCI, apparent areas of cavitation activity decreased when products of the emissions were used to perform frequency-sum beamforming. However, frequency-sum beamforming also produced artifacts, including the appearance of spurious emission sources.

2:45

3pBA8. Estimation of damping coefficient based on the impulse response of echogenic liposomes. Jason L. Raymond (Dept. of Biomedical Eng., ThoraxCtr., Erasmus MC, CVC 3940, 231 Albert Sabin Way, Cincinnati, OH 45267-0586, raymonjl@mail.uc.edu), Ying Luan, Tom van Rooij (Dept. of Biomedical Eng., ThoraxCtr., Erasmus MC, Rotterdam, Netherlands), Shao-Ling Huang, David D. McPherson (Dept. of Internal Medicine, Univ. of Texas Health Sci. Ctr., Houston, TX), Nico de Jong (Dept. of Biomedical Eng., ThoraxCtr., Erasmus MC, Netherlands, Netherlands), and Christy K. Holland (Dept. of Internal Medicine, Univ. of Cincinnati, Cincinnati, OH)

Echogenic liposomes (ELIP) are under development as theragnostic ultrasound contrast agents for the diagnosis and treatment of cardiovascular disease. ELIP formulations have a phospholipid bilayer shell and are echogenic due to the presence of air; however, the exact location of the entrapped air has not been fully ascertained. Air pockets could either be stabilized by lipid monolayers within the liposome, or by the lipid bilayer shell. Our goal is to develop a more complete understanding of the encapsulation and shell properties of ELIP. This study demonstrates a method to estimate the damping coefficient using experimentally measured radius-time curves of the impulse response of individual ELIP using optical methods. The non-dimensional damping coefficient and the natural frequency of oscillation were estimated based on 140 individual impulse responses as measured with the Brandaris 128 fast-framing camera (15 Mfps) at 37°C. The damping coefficient was in agreement with the damping coefficient as measured previously using a broadband acoustic technique [Raymond *et al.*, Ultrasound Med Biol. **40**(2), 410–421 (2014)]. However, the natural frequency of oscillation was lower than previously reported.

3:00

3pBA9. The stable nonlinear acoustic response of free-floating lipid-coated microbubbles. Ying Luan, Guillaume Renaud, Tom Kokhuis, Antonius van der Steen, and Nico de Jong (Biomedical Eng., Erasmus Medical Ctr., Pieter de Hoochweg 119A, Rotterdam 3024 BG, Netherlands, y.luan@erasmusmc.nl)

The stability of the microbubbles maintained by the lipid coating is crucial for diagnostic contrast-enhanced ultrasound imaging. We present a study of the stability of the dynamic response of single free-floating microbubbles (DSPC-based homemade microbubbles) with an acoustical camera. A 30 MHz probing wave (800 μs duration) measures the dynamic response of single microbubbles to 42 short sine bursts (1 MHz, 10 μs duration, 3 μs interval between each excitation) at three different peak pressures (25, 100,

and 200 kPa). For each microbubble exposed to the 42 consecutive pulses, the following parameters were calculated: the radial strain at the driving frequency (ε_f), at the second/third harmonic frequencies (ε_{2f},ε_{3f}), the ratio of radial excursion in expansion over that in compression (EoC) and the dc offset in the time-domain response. Nearly all 1500 individual microbubbles measured showed stable vibrational response. As expected ε_{2f} and ε_{3f} increase with ε_f, but they reach a plateau when ε_f exceeds about 30%. For ε_f smaller than 15%, we observed compression-dominant behaviors (dc offset < 0 and EoC < 1), while microbubbles show expansion-dominant responses (dc offset > 0 and EoC > 1) for ε_f larger than 15%.

WEDNESDAY AFTERNOON, 7 MAY 2014 557, 1:30 P.M. TO 2:50 P.M.

Session 3pID

Interdisciplinary: Hot Topics in Acoustics

Tessa Bent, Chair
Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405

Chair's Introduction—1:30

Invited Papers

1:35

3pID1. Hot Topics—Hidden hearing loss: Permanent cochlear-nerve degeneration after temporary noise-induced threshold shift. M. Charles Liberman and Sharon G. Kujawa (Eaton Peabody Labs., Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston, MA 02114, charles_liberman@meei.harvard.edu)

The classic view of sensorineural hearing loss (SNHL) is that the "primary" targets are hair cells, and that cochlear-nerve loss is "secondary" to hair cell degeneration. Our work in mouse and guinea pig has challenged that view. In noise-induced hearing loss, exposures causing only reversible threshold shifts (and no hair cell loss) nevertheless cause permanent loss of >50% of cochlear-nerve/hair-cell synapses. Similarly, in age-related hearing loss, degeneration of cochlear synapses precedes both hair cell loss and threshold elevation. This primary neural degeneration has remained hidden for two reasons: (1) the spiral ganglion cells, the cochlear neural elements commonly assessed in studies of SNHL, survive for years despite loss of synaptic connection with hair cells, and (2) the degeneration is selective for cochlear-nerve fibers with high thresholds. Although not required for threshold detection in quiet (e.g., threshold audiometry, auditory brainstem response threshold), these high-threshold fibers are critical for hearing in noisy environments. Our research suggests that (1) primary neural degeneration is an important contributor to the perceptual handicap in SNHL, and (2) noise exposure guidelines should be re-evaluated, as they are based on the faulty premise that threshold audiograms are the most sensitive measures of noise damage.

2:00

3pID2. Energy harvesting from acoustic fields. Kenneth Cunefare (Georgia Tech, Mech. Eng., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu)

While energy harvesting from a variety of ambient sources (vibration, light, and wind) has been demonstrated and sensing and communication applications to exploit those sources have been developed, acoustic energy as an ambient energy source has not received similar attention, except for a few very specialized special cases; this "hot topic" will focus on the special cases. The reason for otherwise limited development comes down to the basic physics of how much energy is available within a "typical" acoustic field. For airborne sounds, the energy density in sound fields that are perceived by humans to be loud to painfully loud (e.g., 80 to 140 dB, or ~0.2 Pa to ~200 Pa) actually represent an extremely low available energy source. In consequence, means must be taken to intensify an acoustic response, for example through resonance, but even so, the available energy remains limited. But, what if the sound field is not "typical"? One of the exceptions which enables viable acoustic energy harvesting is the sound field that exists inside of an operating jet aircraft engine. Another exception is within pumped and pressurized fluid systems, where acoustic pressure variations may reach into the mega-Pascal (MPa) range. Energy harvesting from such a fluid-borne acoustic source is feasible for powering sensors and wireless communication systems, has been successfully demonstrated, and may yield commercialized technology within only a few years.

2:25

3pID3. Underwater sound from pile driving, what is it and why does it matter. Peter H. Dahl, Per G. Reinhall (Appl. Phys. Lab. and Dept. of Mech. Eng., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, dahl@apl.washington.edu), Arthur N. Popper (Dept. of Biology, Univ. of Maryland, College Park, MD), Mardi C. Hastings (Dept. of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), and Michael A. Ainslie (Acoust. and Sonar Group, TNO, The Hague, Netherlands)

Pile driving as used for in-water construction can produce high levels of underwater sound that has potential to produce physiological and/or behavioral effects on fish, benthic invertebrates, and marine mammals. There are two basic pile driving methods: impact pile driving where the pile is driven by strikes from a high-energy hammer, and vibratory pile driving where the pile is effectively vibrated into the sediment. Often both methods are used on the same pile. At ranges on the order of 10 m, and considering steel piles of diameter 0.75–1 m, vibratory pile driving produces underwater sound pressures of order 100–1000 Pa, which is often sustained for minutes. In contrast, each impact pile strike produces peak sound pressures on the order of 100 kPa, with effective duration of the sound from the strike being of order tens of milliseconds. Measurements made both far from the pile source (range many depths) and close-in (range of about 1–2 depths) for impact and vibratory pile driving are presented, along with examples modeling of such sound. We conclude by showing why such sounds matter to aquatic life; potential effects include injury at close range and behavioral changes, including evasion resulting in habitat loss at greater distance.

WEDNESDAY AFTERNOON, 7 MAY 2014 — 551 A/B, 12:55 P.M. TO 2:45 P.M.

Session 3pPA

Physical Acoustics: Topics in Nonlinear Acoustics

Dimitri Donskoy, Chair
Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030

Chair's Introduction—12:55

Contributed Papers

1:00

3pPA1. The phenomenon of self-trapping of a strongly nonlinear wave. Oleg Rudenko (Acoust., Moscow and Nizhni Novgorod Univ., General Phys. and Earth Phys. Inst. of RAS, Moscow State Univ., Moscow, Russian Federation, rudenko@acs366.phys.msu.ru) and Claes Hedberg (Eng. Sci., Blekinge Inst. of Technol., Karlskrona, Sweden)

Self means here an effect of a wave on itself. Several self-action phenomena are known in nonlinear wave physics. Among them are self-focusing of beams, self-compression of light pulses, self-channeling, self-reflection (or self-splitting) waves with shock fronts, self-induced transparency, and self-modulation. These phenomena are known for weakly nonlinear waves of different physical origin. Our presentation at ASA meeting in Montreal [POMA **19**, 045080 (2013)] was devoted to strongly nonlinear waves having no transition to the linear limit at infinitesimally small amplitudes. Such waves can demonstrate particle-like properties. Self-trapping consists of the arrest of wave propagation and in the formation of a localized state. In particular, the model generalizing the Heisenberg' ordinary differential equation to spatially distributed systems predicts periodic oscillations but no traveling waves. Different models for strongly nonlinear waves will be considered and some unusual phenomena will be discussed. Preliminary results were published in Ac. Phys. **59**, 584 (2013) and Physics-Uspekhi (Adv. Phys. Sci.) **183**, 683 (2013). [This work was supported by the Megagrant No.11.G34.31.066 (Russia) and the KK Foundation (Sweden).]

1:15

3pPA2. Nonlinear acoustic waves in media with hysteresis and longtime relaxation. Lev A. Ostrovsky (PSD, NOAA ESRL, 325 Broadway, R/PSD99, Boulder, CO 80305, lev.a.ostrovsky@noaa.gov)

It is known that the media with complex structures (e.g., rocks and ceramics) possess an anomalously strong elastic nonlinearity. Nonlinear acoustic effects are used in a number of important applications, such as seismic waves (earthquakes), non-destructive testing, and contact physics. Two specific features were registered, in most of the experiments: hysteresis in the stress-strain relation and longtime relaxation (slow time). Some physical models, mostly phenomenological, have been suggested to explain these phenomena. However, there are very few works considering the effect of medium hysteresis and relaxation on wave propagation and oscillations in resonators. This presentation is a review of both published and new results in this area. Among the problems discussed are: (1) A brief overview of experiments and models; (2) Analysis of wave propagation in simplified models of media with hysteresis as opposed to the nonlinear waves in media characterized by elastic constants of second and third order; (3) Theory of wave interaction in resonators with hysteresis; (4) Study of waves in media with slow time relaxation; (5) Some data from laboratory and field experiments; (6) Some unsolved issues and the relevant future work will be outlined in conclusion.

1:30

3pPA3. Development of high intensity focused ultrasound transducers to deliver specified shock wave amplitudes at the focus. Vera A. Khokhlova, Petr V. Yuldashev, Pavel B. Rosnitskiy (Phys. Faculty, Moscow State Univ., Leninskie Gory, Moscow, Russian Federation, vera@apl.washington.edu), Adam D. Maxwell (Dept. of Urology, Univ. of Washington School of Medicine, Seattle, WA), Wayne Kreider (Ctr. for Industrial and Medical Ultrasound, Univ. of Washington, Seattle, WA), and Oleg A. Sapozhnikov (Phys. Faculty, Moscow State Univ., Moscow, Russian Federation)

High intensity focused ultrasound (HIFU) is currently emerging into many clinical applications. Certain HIFU modalities, histotripsy, for example, rely on the formation of high amplitude shocks in the focal waveform of the beam. These shocks develop at different focal pressure levels depending on the geometry of the HIFU transducer. The goal here was to determine optimal transducer parameters that would result in specified shock amplitudes and corresponding peak negative focal pressures. The hypothesis was that pressure level for shocks to form is mainly determined by the F-number of the transducer. As nonlinear effects accumulate almost entirely in the focal lobe of HIFU beams, shocks will form at the higher pressures for lower F-number transducers with shorter focal lobe and thus will have higher amplitudes. Simulations based on the Khokhlov-Zabolotskaya-Kuznetsov equation have shown that for typical HIFU transducers with 1–3 MHz frequencies, geometries with F-number close to 1 are optimal for generating waveforms with about 70 MPa shocks and 12 MPa peak negative pressures. For lower F-number transducers, higher amplitude shocks and peak negative pressures will be formed unless cavitation occurs proximal to the focus to attenuate the focal beam. [Work supported by the grants MK-5895.2013.2, RFBR-13-02-0018, NIH-EB007643, and T32-DK007779.]

1:45

3pPA4. Nonlinear collapse of vortex. Konstantin A. Naugolnykh (Phys., Environ. Study Res. Lab., NOAA, 325 Broadway, Boulder, CO 80305, konstantin.naugolnykh@noaa.gov)

Sound radiation by two vortices with different intensity and sign changes the distribution of vortex field. The acoustic instability and vortices motion may occur in such a system. The linear stage of this process was considered previously the spread of similar vortices (Klyatskin, 1966) and attraction of different vortices (Gryanik, 1983; Kop'ev and leont'ev, 1983). As a result of attraction, the nonlinear effect of collapse of vortices may appear. This process is considered by the method of matched asymptotic expansion of the solution for the point vortices in an incompressible fluid and the solution of nonlinear Burgers equation. The features of cylindrical wave spread and nonlinear distortion both indicated.

2:00

3pPA5. Enhancement of a general solution to Lighthill-Westervelt nonlinear acoustic equation to the cases of inhomogeneous and random media. Harvey C. Woodsum (Hobbit Wave, Inc., 21 Continental Blvd., Merrimack, NH 03054, hwhwoodsum@gmail.com)

A general solution to the Lighthill-Westervelt equation of nonlinear acoustics, previously developed and successfully applied to model the scattering of sound by sound and to the parametric array, has been generalized further for use in the cases of inhomogeneous and random media. The form of the solution makes use of an exact inverse differential operator in combination with a sequence of perturbation terms that comprise the multiple orders of nonlinear acoustic scattering to arbitrary order. Integration techniques have been developed which allow accurate, approximate, analytical solutions under particular circumstances. These solutions are shown to reduce to other previously known solutions in the appropriate limits.

2:15

3pPA6. Analysis of scattering from dual-frequency Incident beam interaction. Chrisna Nguon (Univ. of Massachusetts Lowell, 63 Hemlock St., Dracut, MA 01826, chrisna_Nguon@student.uml.edu), Nicholas Misiunas, Barbara Deschamp, Kavitha Chandra, and Charles Thompson (Univ. of Massachusetts Lowell, Lowell, MA)

The characterization of acoustically inhomogeneous structures insonified by dual-frequency acoustic beams is undertaken. A layer of microbubble contrast agents is included to improve the resolution of the scattered field. Of particular interest is the identification of characteristic sizes of volume scatterers and their principal orientations by measurement of their acoustic signatures in the exterior field. The inclusion of bubble contrast agents has shown to enhance the directional pattern of the back-scattered field generated from the difference frequency source created by the nonlinear interaction of the incident beams. However, the sensitivity of the scattered field to variations in the contrast parameter and spatial orientation requires computation to be carried out in a high-dimensional parameter space that includes the difference wavenumber, angle of incidence, acoustic nonlinearity parameter, and several microbubble parameters. A fast computation of the exterior field is carried out using a monopole expansion of the Green's function that separates the source and observation coordinates. The exterior field is sampled and analyzed in the aforementioned system parameter space to identify the weighted combination of parameters that serve as a minimum representation for detecting the changes in the medium parameters through measurements of the exterior scattered field.

2:30

3pPA7. Analysis of strongly nonlinear blast waves using the Rankine-Hugoniot relations and the nonlinear ray theory. Jae-Wan Lee, Won-Suk Ohm (School of Mech. Eng., Yonsei Univ., Yonsei University, 50 Yonsei-ro, Seodaemun-gu, Seoul 120749, South Korea, fantalex@yonsei.ac.kr), and Woosup Shim (The 5th R&D Institue-2, Agency for Defense Development, Daejeon, Daejeon, South Korea)

Blast waves produced by such events as nuclear explosions can be considered as strongly-nonlinear shock waves, the description of which requires a theoretical framework more accurate than the second-order wave equation. In this regard, computational fluid dynamics (CFD) techniques based on the Euler equation are frequently used. However, CFD techniques are very time-consuming for blast waves traveling over great distances. This paper presents a theoretical framework for propagation of strongly nonlinear blast waves, which shines in both speed and accuracy. Local propagation speeds of a blast wave are obtained by applying the Rankine-Hugoniot relations to "infinitesimal shocks" between adjacent phase points comprising the blast wave. By grafting the propagation speed onto the nonlinear ray theory, the evolution of the blast wave can be computed. Phenomena characteristic of strongly-nonlinear blast waves such as the Mach reflection, the Mach stem generation, and the self-refraction are observed from numerical simulations.

Session 3pSA

Structural Acoustics and Vibration, Noise, and Architectural Acoustics: Acoustics of Sports

Matthew D. Shaw, Cochair
Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802

Donald B. Bliss, Cochair
Mech. Eng., Duke Univ., 148B Hudson Hall, Durham, NC 27705

Chair's Introduction—1:25

Invited Papers

1:30

3pSA1. Vibrational assessment of wood, composite, and plastic hurleys. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu)

The Gaelic sport of hurling combines elements of field and ice hockey, lacrosse, and even baseball. The hurling stick, or hurley, has a long narrow handle which tapers to a large flat paddle, or bas, and is primarily made from white ash. Sticks show variation in the thickness, size, and shape of the paddle and composite hurleys have recently been introduced to the game. In this paper, we use experimental modal analysis to study the vibrational mode shapes and frequencies of several ash and composite hurleys, in adult and youth sizes, including the infamous 1970's Wavin plastic hurley which was quickly abandoned due to excessive vibration and sting. Bending and torsional mode shapes are found to be similar to those in baseball bats and field hockey sticks. A third family of vibrational modes exhibiting bending in the handle and torsion in the paddle are similar to vibrational modes observed in ice and field hockey sticks. These three types of mode shapes help define the sweet spot as well as influencing the perception of feel in the hands of a player.

1:50

3pSA2. Remotely monitoring performance in sports acoustically. David Browning (139 Old North Rd., Kingston, RI 02881, decibeldb@aol.com) and Peter Scheifele (Dept. of Comm. Sci., Univ. of Cincinnati, Cincinnati, OH)

In many sports, a coach will tell you that their trained ear can detect a superior performance. Volleyballs, soccerballs, and footballs all resonate at a characteristic frequency when struck. The resulting sound can be monitored off the field of play to determine how hard they were hit. A simple sound level meter can be easily modified to make a smackmeter. A bit more complicated but perhaps more rewarding is the resonate sound from aluminum baseball bats, or tennis and squash reckets, when can reveal not only how well but where the hit was made. Monitoring the sound of the stumming shaft of a golf club during a swing gives valuable information on swing speed and uniformity. Most every sport, for example, how about the splash of a dive, appears to have sounds that could be worthwhile to remotely monitor, especially given the resolution and speed of modern analysis techniques.

2:10

3pSA3. Racquetball exposed: Analysis of decay times and exposure levels in racquetball courts. Matthew D. Shaw and Eric C. Mitchell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mdshaw16@gmail.com)

Racquetball courts are interesting acoustic spaces. The architecture varies from court to court; therefore, the acoustic environment also changes. This talk will present reverberation time measurements of two types of courts: fully enclosed courts and courts with open viewing areas. Similarities and differences in the decay times will be discussed. This research will also investigate a player's exposure level during a typical racquetball match.

Contributed Papers

2:30

3pSA4. Sound transmission issues in higher-story fitness facilities. Sharon Paley (ECORE Int., 715 Fountain Ave., Lancaster, PA 17601, sharon.paley@ecoreintl.com)

The growing popularity of Crossfit and other high-intensity workouts coupled with an increasing trend for gyms/weight rooms to be installed on higher stories of lighter-weight buildings is casting a new light on sound transmission issues in fitness facilities for the acoustic consultant. Traditional gym flooring and mats do not sufficiently minimize the impact of a dropped dumbbell for the tenant below. This is a growing problem for both residential gyms found in multi-family housing as well as the more extreme facilities for professional athletes and olympians. Field test results, limitations of existing field measurement techniques, and potential solutions will be discussed.

2:45

3pSA5. Sound and noise in high schools gymnasiums. Antonio P. Carvalho and Carla C. Barreira (Lab. of Acoust., College of Eng. (FEUP), Univ. of Porto, FEUP (NIF501413197), R. Dr. Roberto Frias, Porto 4200-465, Portugal, carvalho@fe.up.pt)

The goal of this research was to characterize the interior acoustics of high schools sports facilities using objective parameters. In situ measurements were done in 68 school gymnasiums in Portugal (volume from 450 to 2680 m^3) regarding LAeqBN (background noise without gym classes), LAeqPE (ongoing Physical Education classes), RT, and RASTI. The results for LAeqBN were from 34 dB (L90) to 50 dB (L10) with a median of 42 dB. For the LAeqPE were found values from 75 dB (L90) to 85 dB (L10) with a median of 80 dB. For the RT(500/1k/2k) room values from 2.6 s (RT_{90}%) to 6.9 s (RT_{10}%) with a median of 4.8 s, were measured. The room average RASTI values were from 0.27 ($RASTI_{90}$%) to 0.43 ($RASTI_{10}$%) with a median of 0.34. These sports rooms proved to be highly reverberant, almost without sound absorbing materials, which might be harmful, especially for the gym teachers. The subjective perception of the PE teachers was analyzed through questionnaires where it was verified that they feel most discomfort when it comes for noise (and thermal) conditions. This was supported by the objective results obtained. Ideal values for those acoustic parameters are presented.

WEDNESDAY AFTERNOON, 7 MAY 2014 BALLROOM D, 1:00 P.M. TO 3:00 P.M.

Session 3pSC

Speech Communication: Developmental Topics in Speech Communication

Linda Polka, Chair
School of Commun. Sci. and Disorder, McGill Univ., 1266 Pine Ave., West, Montreal, QC H3G 1A8, Canada

Contributed Papers

1:00

3pSC1. Pre-babbling infants prefer listening to infant speech: Implications for vocal learning in humans. Matthew Masapollo, Linda Polka (McGill Univ., 1266 Pine Ave. West, Montreal, QC H3G 1A8, Canada, matthew.masapollo@mail.mcgill.ca), and Lucie Ménard (Univ. of PQ at Montreal, Montreal, QC, Canada)

For human infants to engage in vocal learning, they must effectively monitor and assess their own self-produced speech, which entails perceiving speech produced by an infant. Yet, little is known about how infants respond to infant-produced speech. Here, we demonstrate that pre-babbling infants prefer listening to infant speech. Across four experiments, 3-to-6-month-olds were tested in a preferential listening procedure, using vowels synthesized to emulate productions by female adults and infants. In experiment 1, infants listened longer to vowels produced by infant than adult speakers. However, in experiment 2, infants failed to show any listening preference for infant versus adult vowels synthesized with matching, infant-appropriate pitch values, suggesting that infants were either attracted to higher voice pitch *per se* or to infant-like voice pitch. Failing to support a bias of the first type, infants in experiment 3 showed no listening preference when presented infant vowels with higher and lower infant-appropriate pitch values. Moreover, in experiment 4, infants showed a preference for infant versus adult vowels when synthesized with pitch values that are appropriate for a female using adult-directed speech; this suggests that infants are also attracted to infant vocal resonance properties. The implications of these results for speech development are discussed.

1:15

3pSC2. Relating moms Apostrophe productions of infant directed speech with their babies Apostrophe ability to discriminate speech: A brain measure study with monolingual and bilingual infants. Adrian Garcia-Sierra (I-LABS, Univ. of Washington, 850 Bolton Rd., Unit 1085, Storrs, Connecticut 06269, adrian.garcia-sierra@uconn.edu), Nairan Ramirez-Esparza (Psych. & Speech, Lang. and Hearing Sci., Univ. of Connecticut, Storrs, CT), Melanie S. Fish, and Patricia K. Kuhl (I-LABS, Univ. of Washington, Seattle, WA)

We report the benefit of Infant Directed Speech (IDS) on speech discrimination in 11- and 14 month-old monolingual (N = 16) and bilingual infants (N = 14). Mothers were instructed to read a booklet to their infants that contained sentences with target words (e.g., park, tear, bark, deer, etc.) at least once a day for four days. This activity was recorded by the LENA digital recorder that the infants were carrying around as they went about their lives. IDS was defined as the mothers' voice-onset time (VOT) durations when reading the booklet at home to their infants. Adult Directed Speech Mothers' IDS productions were correlated with their infants' brain responses associated with speech discrimination (Mismatch Negativity Response or MMR). The results showed that mothers' IDS correlated positively with the bilingual infants' positive-MMR and the monolingual infants' negative-MMR. Bilinguals' positive-MMR is interpreted as a less mature brain response since the positivity declines with age and a negative-MMR (adult-like) emerges later in development. These results show that even though the monolinguals and the bilinguals are at different developmental stages—as demonstrated by their MMRs— both monolingual and bilingual infants benefit from IDS to learn the sounds of their native language(s).

1:30

3pSC3. Infant-directed speech reduces English-learning infants preference for strong/weak versus weak/strong words. Derek Houston and Crystal Spann (Otolaryngol. - Head & Neck Surgery, Indiana Univ. School of Medicine, 699 Riley Hospital Dr., RR044, Indianapolis, IN 46202, dmhousto@indiana.edu)

Mounting evidence suggests that infant-directed speech (IDS) facilitates aspects of language acquisition. An important milestone of English-learning infants' language acquisition is encoding the predominant stress-initial rhythmic structure of English, which 9-month-olds have demonstrated by showing a looking-time preference for lists of strong/weak words (e.g., doctor) versus weak/strong words (e.g., guitar) (Jusczyk *et al*., 1993). We tested for this preference in 48 9-month-olds using the headturn preference procedure. Twenty-four infants were presented with the words using IDS, and 24 were presented with adult-directed speech (ADS). Infants were presented with a visual display of a blinking light on one of two (left and right) monitors. When they oriented toward the monitor, a list was presented from behind that monitor until they looked away for more than 2 s. They were presented with lists of weak/strong words for half the trials and strong/weak words for the other half in quasi-random order. A repeated-measures AVOVA revealed a statistically significant interaction between word-type preference and speech condition. To our surprise, only infants in the ADS condition showed a preference strong/weak words. The findings raise the possibility that rhythmic properties of words may be more difficult for infants to encode in IDS than ADS.

1:45

3pSC4. Differences in the acoustic correlates of intonation in child and adult speech. Jill C. Thorson and James L. Morgan (Dept. of Cognit., Linguistic and Psychol. Sci., Brown Univ., 190 Thayer St., Box 1821, Providence, RI 02912, jill_thorson@brown.edu)

During speech perception, toddlers use prosodic cues, such as pitch, in order to identify the most salient/prominent information in the discourse. This process is critical in facilitating early word recognition and learning. How do children then use these acoustic cues in their own speech in comparison to mature adult speech? The motivation for this study is to examine the acoustic correlates of intonation employed by child (mean: 2.5 years) and adult speakers of English during a guided spontaneous production task. During an interactive game, we elicit a set of target nouns and label them as one of three types: (1) new, uttered for the first time by the participant, (2) given, previously uttered at least once by the participant, or (3) contrastive, uttered in direct opposition to a previously mentioned referent. Along with labeling the pitch accent, we measured f0 range, f0 slope, duration, and intensity for each target word under these varying conditions. These measurements allow us to compare the types of pitch accents used by children and adults, and how each group employs the acoustic correlates of intonation. Identifying how these parameters are used during production is an important step in understanding speech development in early language acquisition.

2:00

3pSC5. Mothers do not enhance phonemic contrasts of Mandarin lexical tones in child-directed speech. Puisan Wong (Speech and Hearing Sci., The Univ. of Hong Kong, 7/F, Meng Wah Complex, Faculty of Education, Pokfulam, Hong Kong, pswResearch@gmail.com), Xin Wang, Wenna Xi, Lingzhi Li, and Xin Yu (The Ohio State Univ., Columbus, OH)

Child-directed speech is characterized by higher pitch, more expanded pitch contours, and more exaggerated phonetic contrasts, which was suggested to facilitate speech sound acquisition. This study examined the perceptual and acoustic differences of mothers' disyllabic Mandarin lexical tones directed to adults and children to determine whether mothers exaggerated the pitch targets of the four Mandarin tones when speaking to children. Twelve Mandarin-speaking mothers produced 700 child-directed (CD) and adult-directed (AD) disyllabic words in a picture naming task. Five Mandarin-native speakers identified the mothers' AD and CD tones in filtered speech. Overall, CD lexical tones were identified with significantly lower accuracy than AD lexical tones (89% vs. 94%, $S = 25.5$, $p = 0.006$, $r = 0.927$, Wilcoxon Signed Rank Test). Acoustic analysis showed that the mean fundamental frequency (f0) of the four tones in both syllables was significantly higher in CD than in AD productions. No difference was found between AD and CD in the distinctive pitch targets for the 4 tones, namely pitch shift for Tone1, F0 slope for Tone2, minimum F0 for Tone3 and F0 slope for Tone4. F0 plots showed mostly parallel contours in AD and CD productions without exaggeration of the phonetic contrasts of the tones. [Work supported by NIDCD F31 DC008470-01A1.]

2:15

3pSC6. Perceptual acuity and production distinctness in child speech: Data from American English /r/. Tara McAllister Byun (Communicative Sci. and Disord., New York Univ., 665 Broadway, New York, NY 10012, tara.byun@nyu.edu) and Mark Tiede (Haskins Labs., New Haven, CT)

Previous work has shown that some adult listeners have more sharply defined perceptual categories than others, and listeners who have the most precise auditory targets also tend to produce more robust contrasts (e.g., Newman, 2003; Perkell *et al*., 2004; Ghosh *et al*., 2010). While it is likely that this relationship also holds in child/adolescent speakers, the hypothesis has not been directly tested. This study compared perception and production of the English /r-w/ contrast in 20 typically developing children aged 9–14. Two 10-step *rake-wake* continua were synthesized using *rake* tokens elicited from two child speakers (8-year-old male, 10-year-old female). Items were presented 8 times each in random order in a forced-choice identification task. Participants also produced *rake* and *wake* in a carrier phrase in casual and careful speech. Perceptual sensitivity was evaluated with a logistic function fitted over the number of *wake* responses at each step in the continuum. Preliminary results show considerable across-participant variation in the slope of the best-fit logistic function, the location of the perceptual boundary, and the acoustic distinctness of /r-w/ in production. We expect to demonstrate a relationship between perceptual acuity and production distinctness similar to that seen in adults. [Work supported by NIH.]

2:30

3pSC7. Timecourse of word recognition for listener-oriented speaking style adaptations. Suzanne V. van der Feest (Commun. Sci. and Disord., Univ. of Texas at Austin, 2504 Whitis Ave. A1100, Austin, TX 78712, suzanne@austin.utexas.edu) and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, Austin, TX)

Previous research has found that listener-oriented speaking style adaptations, such as Infant Directed Speech (IDS) and Clear Speech (CS), aid perception and development and improve intelligibility for children and adults, respectively. However, less is known about whether these different speaking styles enhance intelligibility in general, or if young children benefit most from IDS features aimed at capturing and maintaining attention. Additionally, information about time course differences in word recognition for these speech modifications has not been investigated. This study investigates both questions by presenting young adult listeners with semantically anomalous and meaningful sentences produced in IDS, CS, and Conversational (CO) Speech. Listeners participated in (1) a word-recognition-in-noise test and (2) a visual word recognition paradigm tracking eye movements. Results showed facilitated word recognition in noise for both IDS and CS compared to CO. Both speaking style adjustments also resulted in increased speed of word recognition in the visual paradigm. Semantic context provided additional facilitation only when more exaggerated acoustic cues were present (in IDS and CS). Preliminary analyses show that even for adult listeners IDS provided the greatest perceptual benefits, indicating that established advantages of IDS for young listeners cannot be mainly attributed to affect in addition to enhanced acoustic cues.

2:45–3:00 Panel Discussion

Plenary Session and Awards Ceremony

James H. Miller
President, Acoustical Society of America

Presentation of Certificates to New Fellows

Caroline Abdala - For contributions to understanding the postnatal maturation of the human cochlea, auditory nervous system, and middle ear

Judit Angster - For contributions to the acoustics of the pipe organ

David A. Brown - For contributions to fiber-optic and piezoelectric transduction science, and leadership in acoustics education

John R. Buck - For contributions to applications of random matrix and information theory in acoustic signal processing and bioacoustics

John A. Colosi - For contributions to the science of wave propagation in random media

Huangpin Dai - For contributions to the theory and methodology of the study of central processes in auditory perception

Michael J. Epstein - For integration of physiological and psychological processing in the perception of loudness

Vitalyi E. Gusev - For contributions to nonlinear acoustics and laser-based ultrasonic pulse generation

David R. Schwind - For contributions to the acoustical design of theaters, concert halls, and film studios

Bridget M. Shield - For research, teaching, and leadership to standardize classroom acoustics

Preston S. Wilson - For contributions to the theory and applications of the acoustics of bubbly media

Joseph N. Soker - For contributions to acoustical design and noise control applications in buildings and communities

Presentation of Awards

William and Christine Hartmann Prize in Auditory Neuroscience to Egbert de Boer

Medwin Prize in Acoustical Oceanography to Andone C. Lavery

R. Bruce Lindsay Award to Matthew J. Goupell

Helmholtz-Rayleigh Interdisciplinary Silver Medal to Mark F. Hamilton

Gold Medal to Brian C. J. Moore

Vice-President Gavel to Peter H. Dahl

President's Tuning fork to James H. Miller

Session 3eED

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

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

Cameron T. Vongsawad, Cochair
Phys. & Astronomy, Brigham Young Univ., 1041 E. Briar Ave., Provo, UT 84604

This workshop for Providence area Girl Scouts (ages 12-17) 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 the girls to opportunities in science and engineering and to interact with professionals in many areas of acoustics. A large number of volunteers are needed to make this a success. Please email Traci Neilsen (tnb@byu.edu) if you have time to help with either guiding the girls to the event and helping them get started (5:00 p.m. - 6:00 p.m.) or exploring principles and applications of acoustics with small groups of girls (5:00 p.m. - 7:30 p.m.). We will provide many demonstrations, but feel free to contact us if you would like to bring your own.

WEDNESDAY EVENING, 7 MAY 2014 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, Wednesday, and Thursday evenings.

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 Wednesday are as follows:

7:30 p.m.	Signal Processing in Acoustics	555AB

ACOUSTICAL SOCIETY OF AMERICA

R. BRUCE LINDSAY AWARD

Matthew J. Goupell

2014

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

CITATION FOR MATTHEW J. GOUPELL

. . . for contributions to the understanding of binaural processes in acoustic and electrical hearing

PROVIDENCE, RHODE ISLAND • 7 MAY 2014

Matt Goupell was born in Midland, Michigan in 1979 and grew up in Mt. Pleasant. From an early age, he was inspired by his father's wish that he acquire an education that would encourage him to value what is "right and good." In 2001, Matt received a bachelor's degree from Hope College in Holland, Michigan based on a linear combination of undergraduate physics and Ultimate Frisbee. He immediately began graduate work in nuclear physics at Michigan State University (National Superconducting Cyclotron Laboratory). There, he studied quantum mechanics, wrote C-code for heavy-ion experiments, hauled lead bricks, and did all the other things that nuclear physicists do until he realized how much more rewarding and fun a career in acoustics would be. Exactly how he came to this decision is unclear. Perhaps it stemmed from his love of music (He sang, played the viola and guitar). Perhaps he realized that acousticians are just great people. However it happened, one of us (WMH) had the good fortune to snag Matt into his psychoacoustics lab to study human binaural hearing.

Matt's graduate work consisted of an experimental study of the detection of binaural incoherence near the limit of perfect coherence. In the narrow-band limit, the experimental conditions are similar to the famous NoSpi condition in masking level difference studies, but the emphasis on incoherence detection gave the work important simplifying power as the bandwidth was increased and the duration varied. Matt's experiments demonstrated the primary importance of interaural fluctuations in detection. He tested ten binaural models against his incoherence detection data, which ultimately supported a model based on a combination of separately processed contributions from interaural phase fluctuations and interaural level fluctuations. That modeling effort exhibited several things about Matt's research style. First, it demonstrated a deep grasp of the vast binaural literature. Next, it demonstrated Matt's keen desire to get to the bottom of things and his dogged determination to find out why experiments turn out the way they do. As an experimenter, Matt made a very good theorist. It demonstrated too that it is possible to do science with your leg in a cast resulting from injuries sustained during vigorous matches of Ultimate. Matt played on the Michigan team that competed nationally and also coached the University team. Matt's thesis work led to four research articles in the *Journal of the Acoustical Society of America.* Before (finally) leaving Michigan in 2005, Matt had acquired a Ph.D. and also found a wife, Sarah, another devotee of Ultimate. They had met at a tournament several years previously.

Next, Matt's thirst for paradigm shifts and his creativity took his research career on a translational path. He decided to apply his signal processing and psychoacoustic skills to study how human patients, who use cochlear implants (CIs) and thus perceive sound through electrically pulsed stimulation of the auditory nerve ("electrical hearing"), are able to hear, especially binaurally. This decision took Matt and Sarah to Vienna, where Matt spent three years conducting novel research with Drs. Bernhard Laback and Peter Majdak at the Austrian Academy of Sciences. He made important contributions to the role of temporal jitter in improving sensitivity to interaural time differences for both implantees and normal hearing listeners. The idea that localization precision can be improved by adding randomness to the stimulus is counterintuitive, but Matt and his colleagues showed that the effect occurs not only perceptually in the normal human binaural system but also can be effectively modeled as was verified by measured neural responses in animal brainstem. Matt further drew upon his familiarity with the psychoacoustics literature in showing that a number of effects seen in listeners with normal hearing can also be seen in implantees, if the paradigms are carefully designed to ask the right questions. Over the next five years, he studied sound localization in both horizontal and vertical planes; he applied profile analysis to make a test for implantees; and he showed that implantees can exhibit enhancement effects when signals are pulsed. He also led the Vienna team in Ultimate Frisbee – compiling an enviable record against teams in Italy and Hungary – a bold venture into athletics, tourism, international relations, and gastronomy.

Having been smitten by the applied nature of the CI field, which lends itself to both theoretical and applied questions, Matt decided to return to the United States to delve more deeply into issues of binaural hearing in CI users. In 2009, Matt joined Ruth Litovsky's lab at the University of Wisconsin-Madison for a second postdoctoral opportunity, where he delved into new areas involving speech intelligibility and spatial unmasking. Within a year, he received the relatively new and highly coveted "Pathways to Independence" NIH K99/R00 award. This grant presented an ideal opportunity for Matt to apply his signal processing skills and knowledge of binaural hearing to deeply rooted questions regarding the access that binaural CI users may have to stimulation that normal hearing listeners can process effectively.

Since the fall of 2011 Matt has been heading up his own research program as an assistant professor in the Department of Hearing and Speech Sciences at the University of Maryland in College Park and as a member of the Neuroscience and Cognitive Science Program. The quest for knowledge regarding the effectiveness of CIs continues to be the driving force in his current work. Matt's experiments have important implications for the development of auditory prostheses that enable deaf individuals to hear in noisy environments and to localize sounds. He has addressed vexing problems stemming from limitations in the clinical procedures, such as how to best present patients with signals in the two ears that are loudness-balanced and that produce a reliable auditory image. Since his days at Michigan State, he has published 12 additional articles/letters in the *Journal of the Acoustical Society of America* as well as three articles in other refereed journals. He has not only been productive, he is also highly visible with authorship on 11 talks and 15 posters in 2013 alone! Matt's development has seen him become an effective teacher, valued reviewer, kind mentor, and a father. He and Sarah have two sons, Caleb (4) and Andrew (1). He is also an active member of the Acoustical Society of America, an inspiring invited speaker, and a closet expert in Chinese cooking. All this has come at a price, however. Matt has apparently given up Ultimate Frisbee in favor of back alley basketball with other dads in the neighborhood.

We are delighted to congratulate Matt on behalf of his many colleagues, friends, students and collaborators on being awarded the R. Bruce Lindsay Award.

RUTH Y. LITOVSKY
WILLIAM M. HARTMANN

ACOUSTICAL SOCIETY OF AMERICA
HELMHOLTZ-RAYLEIGH INTERDISCIPLINARY
SILVER MEDAL
in

Physical Acoustics and Biomedical Acoustics

Mark F. Hamilton

2014

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	Gilles A. Daigle	2005
David E. Weston	1998	Mathias Fink	2006
Jens P. Blauert	1999	Edwin L. Carstensen	2007
Lawrence A. Crum	2000	James V. Candy	2008
William M. Hartmann	2001	Ronald A. Roy	2010
Arthur B. Baggeroer	2002	James E. Barger	2011
David Lubman	2004	Timothy J. Leighton	2013

Interdisciplinary Silver Medal

Eugen J. Skudrzyk	1983
Wesley L. Nyborg	1990
W. Dixon Ward	1991
Victor C. Anderson	1992
Steven L. Garrett	1993

CITATION FOR MARK F. HAMILTON

. . . For contributions to nonlinear acoustics and biomedical ultrasound

PROVIDENCE, RHODE ISLAND • 7 MAY 2014

The Acoustical Society awards Mark Hamilton its Helmholtz-Rayleigh Interdisciplinary Silver Medal for his extensive research in nonlinear acoustics, his development of more realistic bubble physics for biomedical applications, and for work on applications that contribute to engineering acoustics.

Mark Francis Hamilton was born in Holyoke, Massachusetts on August 7, 1956. His family moved to St. Louis and later to Brussels, Belgium, where Mark spent his high school years. After the family returned to Holyoke, Mark entered Columbia University in 1974 to study electrical engineering. Two senior-year courses under Cyril Harris pointed Mark toward acoustics, and with Cyril's encouragement, Mark entered the graduate acoustics program at Pennsylvania State University in June 1978. Experimental research at the Applied Research Laboratory's Garfield Thomas Water Tunnel on cavitation bubble noise led to a master's degree. However, Mark found his calling when he took Frank Fenlon's Nonlinear Acoustics course, and in 1980 he began doctoral research on dispersion effects in parametric arrays under Frank's supervision.

Although Mark was fascinated by nonlinear acoustics, the way was not easy. Frank Fenlon died of cancer in June 1981, midway though Mark's research. Frank and David Blackstock had been longtime friends; in fact, at Frank's instigation, Mark and David had met a month earlier, at the ASA Ottawa Meeting. Because none of its other faculty members were working in nonlinear acoustics at the time, Penn State arranged for David, at Applied Research Laboratories, University of Texas at Austin (ARL:UT), to take Frank's place as Mark's supervisor. The topic remained the same, but Mark moved to ARL:UT to finish the work in absentia. Mark took his final exams and defended his dissertation at Penn State in June 1983.

While at ARL:UT Mark met and worked with Sigve and Jacqueline Naze Tjøtta, visiting scientists from the University of Bergen (Norway). When Mark was awarded ASA's F. V. Hunt Postdoctoral Research Fellowship in 1983, he spent the fellowship year with the Tjøttas in Bergen. Thus was deepened the collaboration with the Tjøttas, which extended over several years and was very fruitful, particularly on finite-amplitude sound beams, a broad field in which Mark would make major contributions for the next 20 years.

On completion of his Hunt Fellowship, Mark returned to Texas as a research fellow at ARL:UT, and in 1985 he began his teaching career as Assistant Professor in the Department of Mechanical Engineering. One of his earliest projects was on focused finite-amplitude beams. Although the resulting 1988 paper seemed innocuous at the time, it turned out to be a very popular work as interest exploded in focused high-intensity sources for applications in medical ultrasound. Related work, on transient and pulsed beams, led to a much more efficient way of solving the KZK equation, a widely used model for problems in which nonlinearity, absorption, and diffraction are important. Along the way, Mark published a benchmark small-signal solution for the axial field of a spark-source lithotripter used to disintegrate kidney stones ("Transient Axial Solution for the Reflection of a Spherical Wave from a Concave Ellipsoidal Mirror," 1993).

Meeting Evgenia (Zhenia) Zabolotskaya—the "Z" in KZK—and Yurii Ilinskii on a visit to the former Soviet Union in 1987 led to another extremely fruitful collaboration. After several exchange visits, Zhenia and Yura joined Mark's research group permanently in 1991. They developed and explored a completely new field of nonlinear acoustics: nonlinear shear and surface waves, for example, Rayleigh waves.

Although the KZK equation was key to expanding nonlinear acoustics into the field of directional radiation produced by common sources, such as sonar and loudspeakers, the equation was very difficult to solve. Success was largely limited to very specific problems, for example, radiation from monofrequency or bifrequency sources. Mark found a way to deal with directional sources that emit broadband signals, such as pulsed piston radiation (commonly used in biomedical ultrasound), other transients, and noise (Lee and Hamilton, "Time-Domain Modeling of Pulsed Finite-Amplitude Sound Beams," 1995). Later, he and Robin Cleveland extended the method to relaxing fluids (Cleveland, Hamilton, and Blackstock, "Time-Domain Modeling of Finite-Amplitude Sound in Relaxing Fluids," 1996) so that, for example, sonic boom propagation in the atmosphere or biomedical ultrasound beams in tissue, could be treated.

About a decade ago, with Zhenia and Yura, Mark turned his attention to bubble physics, particularly with application to biomedical ultrasound. His research group wanted to go beyond the classical problem of a single bubble in an infinite liquid, too ideal a model for many practical problems, for example, bubbles in blood vessels, bubbles in biological tissue, and bubble clouds produced by lithotripsy in the kidney. Mark and his research group opened up the analysis to cover multiple bubbles and their interactions, effects of nearby surfaces on cavitation, and more realistic properties of the host medium, such as viscoelasticity. The outpouring of results shows Mark to be at the forefront of practical applications of bubble physics and biomedical ultrasound in general.

Mark's research has helped implement several applications in engineering acoustics, for example, parametric array technology (in water, air, and soil), thermoacoustics, and, more recently, noise underwater produced by offshore wind turbine farms.

The Acoustical Society of America (ASA) has been at the center of Mark's professional life. He has served on countless ASA committees, many as chair, and served as Member of the Executive Council, Vice President, and President. He is currently an Associate Editor of the Journal of the Acoustical Society of America (JASA) as well as JASA Express Letters. Mark's service has extended beyond the ASA. He was appointed as one of ASA's representatives to the Governing Board of the American Institute of Physics (AIP), where for 6 years he was an outspoken advocate for the ASA, and for science in general, as a member of several AIP committees. He is also very active in international acoustics. Long having a leadership role in the International Symposia on Nonlinear Acoustics (ISNA), he has for many years been its General Secretary. Finally, he has recently been appointed to the Board of the International Commission for Acoustics as U.S. representative.

Mark's early promise was shown by his selection to be the Hunt Postdoctoral Fellow in 1983. The promise blossomed, and he received the Lindsay Award in 1989. Now, in recognition of his full maturity and mastery of several fields, the Acoustical Society honors him with its Helmholtz-Rayleigh Interdisciplinary Silver Medal.

DAVID T. BLACKSTOCK
LAWRENCE A. CRUM
GARY W. ELKO

ACOUSTICAL SOCIETY OF AMERICA

GOLD MEDAL

Brian C. J. Moore

2014

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

CITATION FOR BRIAN C. J. MOORE

. . . for leadership in research on human hearing and its clinical applications

PROVIDENCE, RHODE ISLAND • 7 MAY 2014

Brian Moore's name has been synonymous with psychoacoustics for several decades. Having studied Natural Sciences at the University of Cambridge, and completed a Ph.D. there in 1971, it took him only 6 years to be recruited back to Cambridge as a faculty member, where he was promoted to his current position as Professor of Auditory Perception in 1995 and where he continues to reside with his wife and fellow psychoacoustician, Hedwig Gockel.

His textbook "An Introduction to the Psychology of Hearing," published in 1977 and now in its sixth edition, has been an exciting first step into the world of auditory perception for generations of students around the world, and remains an important reference for researchers throughout their careers. Browsing through the chapter titles, it is easy to see why Brian has become such an icon in the field: from frequency selectivity and masking, through loudness, pitch, auditory pattern and object perception, to practical applications, there is scarcely an area to which Brian has not made substantial and seminal contributions.

Frequency selectivity is established in the cochlea of the inner ear, and has profound effects on practically all aspects of auditory perception. Beginning in the late 1970s, Brian and his team produced a series of elegant experiments that devised and refined behavioral methods to probe the frequency selectivity of the human auditory system, using masking techniques to systematically avoid many of the artifacts and confounds that had plagued earlier attempts. Brian refined the notched-noise technique, pioneered by his long-time Cambridge colleague, Roy Patterson, to provide a "map" of frequency selectivity over the audible range of frequencies. This 1990 study, published with long-time lab associate Brian Glasberg, has become the classic reference for estimating human auditory frequency tuning and has been cited more than a thousand times.

Loudness is one of the primary perceptual attributes of sound, and here again Brian has made some of the most lasting contributions to our understanding of how loudness relates to peripheral auditory processing, and how loudness is affected by hearing loss. His computational models of loudness have been met with near-universal acceptance (a rarity in this field), and have found application in the design of hearing aids and in noise assessment and abatement.

Pitch is another fundamental aspect of auditory perception and again Brian has shaped research in this area since his earliest work in the 1970s through his most recent contributions to the ongoing debate on the role of temporal fine structure. Combining ingenious experimental design with computational modeling, he has helped to answer long-standing questions about the nature of pitch perception and its neural bases.

A constant theme running through his career has been the application of his research to the theory, diagnosis, and treatment of hearing loss. His theoretical contributions include demonstrations of how the reduced compression in the impaired cochlea influences loudness and temporal resolution in people with sensory hearing loss, and how a reduction in frequency selectivity affects how these people hear the pitch of both simple and complex sounds. His practical contributions have been equally diverse and influential, including a widely adopted method for fitting hearing aids, an effective automatic-gain control system that has been incorporated in a commercial cochlear implant, and the development of a test that allows audiologists to identify "dead regions" in a patient's cochlea. When evaluating the latest high-tech solutions to hearing impairment, it is perhaps worth reflecting that Brian's achievements have required little more than a computer and a pair of headphones.

Brian is by far the most prolific psychoacoustician of all time with over 480 peer-reviewed journal articles published, about half of them in the *Journal of the Acoustical*

Society of America (JASA). At a time when British university departments were undergoing national research productivity assessments, the point was made, only half-jokingly, that Brian himself would be categorized as an internationally leading department.

Although the Acoustical Society has always been his scientific home, his enormous contributions have not gone unnoticed in the wider scientific world. He has received honors from numerous societies, including the 2008 Award of Merit from the Association for Research in Otolaryngology and the first International Award in Hearing from the American Academy of Audiology. In 2002 he was elected a Fellow of the Royal Society, Britain's highest scientific honor.

Those fortunate enough to have been trained under his mentorship as Ph.D. students or postdocs have known that despite his unrivalled productivity, he is always generous with his time, open to new ideas, and unfailingly enthusiastic in his support of promising young scientists. Scientists from around the world have found it a pleasure to collaborate with Brian; a hallmark of his work is the ability to work easily and successfully with a wide variety of people. The relaxed and friendly atmosphere in his lab is legendary, and in part explains how he has maintained a strong group of senior researchers, Tom Baer, Brian Glasberg, and Michael Stone, in his lab for over 20 years. However, to our knowledge, none of his students or co-workers ever discovered how he maintained his daunting scientific output and submitted detailed and insightful reviews on a near-weekly basis for myriad journals – always on time, if not early – while at the same time always appearing relaxed, having time to play guitar with his jazz band, sing in the college choir, and even play bridge for the college club. Perhaps the secret lies in his passion for fine wines, which he raised to a professional level when he was appointed wine steward of Wolfson College, making him responsible for a substantial annual wine budget, and consequently putting him on very friendly terms with a network of local wine merchants. Regardless of the full answer to this puzzle, the entire community of acoustic science is grateful for the contributions that Brian Moore has made in his distinguished career through his books, his papers, and his personal presence at meetings, where he provides inspiration and encouragement to those around him. The Acoustical Society's Gold Medal is a fitting recognition of his fine achievements.

ANDREW J. OXENHAM
ROBERT P. CARLYON

Session 4aAA

Architectural Acoustics: Green Building Acoustics Design and Challenges

Lucky S. Tsaih, Cochair
Dept. of Architecture, Natl. Taiwan Univ. of Sci. and Technology, Taipei, Taiwan

Gary W. Siebein, Cochair
Architecture, Univ. of Florida, 625 NW 60th St. Ste. C, Gainesville, FL 32607

Chair's Introduction—8:10

Invited Papers

8:15

4aAA1. Noise prediction of vehicle sources on freeways and arterials using measured sound data. John J. LoVerde, David W. Dong, and Samantha Rawlings (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

Evaluation and mitigation of noise from vehicular sources is common as a building design criterion and has been part of the California State Building Code and HUD multi-family building design requirements since the 1970s. It is now included in Green Building Design Standards and school and healthcare facility design guidelines and is expanding to all types of buildings with the addition of the new acoustics credit in LEED version 4. These criteria require that the noise level be quantified precisely, but do not provide a method for defining the noise level given the normal variations in noise level. This paper examines the factors that should be considered when defining the exterior noise from vehicular sources. Methods for predicting the noise level using data from relatively short measurement periods are evaluated, and minimum survey requirements to determine specific exterior noise parameters are suggested.

8:40

4aAA2. Acoustics design associated with natural ventilation. Weihwa Chiang, Huiping Wu, and Haohsiang Hsu (Architecture, National Taiwan Univ. of Sci. and Technol., 43 Keelung Rd., Taipei 106, Taiwan, edchiang1224@hotmail.com)

Increased effort on building sustainability has caused a revolution in building acoustic design regarding savings in resources and energy. Sustainable design strategies such as natural ventilation, on the other hand, challenged the common practice in acoustic design. Case studies on building acoustic solutions associated with natural ventilation have been conducted and reviewed with multi-facet design concerns. Issues addressed included noise insulation of building facade that allows natural ventilation, abatement of noise from urban sub-stations that generated heat, replacing mechanical ventilation system by stack ventilation for parking garages to decrease both power consumption and mechanical noise, and noise suppression of ceiling fan to prevent from overuse of PA system in classrooms that may consequently cause further problems. Discussions were also made about degradation of absorbent materials caused by increased humidity due to natural ventilation.

9:05

4aAA3. Meeting "green" acoustical requirements in flexible classrooms. Rose Mary Su and Benjamin Markham (Acentech Inc., 33 Moulton St., Cambridge, MA 02139, rsu@acentech.com)

Acoustical design in classrooms has evolved significantly since the ANSI S12.60-2002 standard was first established. Since then, institutions such as Leadership in Energy & Environmental Design (LEED) and the Collaborative for High Performance Schools (CHPS) have adopted aspects of the ANSI standard for school projects. Simultaneously, architects are creating increasingly flexible classroom designs. The push for a more flexible learning space sometimes clashes with acoustical design requirements stipulated by LEED and CHPS. This paper will discuss some of the acoustic design challenges of creating flexible, 21st century learning spaces while at the same time meeting the acoustic requirements driven by LEED and CHPS compliance. Discussion will include movable partitions in a learning space that work, sound absorptive finishes implemented beyond the standard suspended acoustical ceiling, and non-conventional mechanical systems in classroom settings. Some case studies will illustrate the discussion.

9:30

4aAA4. Top opportunities and challenges in meeting acoustics criteria in green buildings—Specific case studies. Joseph F. Bridger (Stewart Acoust. Consultants, 7330 Chapel Hill Rd., Ste. 101, Raleigh, NC 27607, joe@sacnc.com)

Acoustics criteria found in for example LEED, DODEA, and ANSI S12.60 (classroom acoustics in schools) often must be met either in addition to or as part of the sustainable design project requirements. The challenge is that building designs are changing to meet the goals of sustainable design, often with unanticipated effects on building acoustics. The criteria themselves are evolving as experience is gained. Acoustics materials are also adapting to meet sustainable design goals. We will discuss the top opportunities and challenges we are finding in meeting acoustics criteria in green buildings.

9:55–10:10 Break

10:10

4aAA5. Leadership in energy and environmental design acoustics compliance: Simulation versus field measurements. Lucky S. Tsaih, Lee-Su Huang, and Lisa Huang (Dept. of Architecture, Natl. Taiwan Univ. of Sci. and Technol., Taipei, Taiwan, akustx@mail.ntust.edu.tw)

In classrooms, acoustics and lighting are equally critical qualities that shape the learning environment. LEED IEQ Prerequisite 3: Minimum Acoustic Performance only addresses a prescriptive requirement for compliance. LEED IEQ Credit 8.1: Daylight and Views allows several options for demonstrating achievement of minimum illumination levels: simulation, prescriptive, measurement, and combination. In examining PK Yonge Elementary School in Gainesville, Florida, field measurements of daylighting and acoustics were performed. Measurements demonstrate that 3 pm illumination levels are better than at 9 am, but model simulations show both to be the same. Model simulation results are at least 10 times better than field measurements. This discrepancy suggests that measurements are critical for more accurate results. Acoustical model simulations for reverberation time were also conducted and the results showed inconsistencies from field measurements. Both daylighting and acoustical simulations provide preliminary results but a range of unpredictable factors affect the final precision of the simulation results. These factors include as-built material finishes, furniture layouts, reflectance, and absorption coefficients. Therefore, LEED should require that upon a building's completion, sample acoustic field measurements are necessary to verify compliance.

10:35

4aAA6. Do elementary school children eat less in noisy cafeterias? Michael Ermann (Architecture + Design, Virginia Tech, 201 Cowgill Hall (0205), Blacksburg, VA 24061-0205, mermann@vt.edu), Elena L. Serrano (Dept. of Human Nutrition, Foods, & Exercise, Virginia Tech, Blacksburg, VA), Carman Byker (Health and Human Development, Montana State Univ., Bozeman, MT), and Robert Calvey (Design for America, Chicago, IL)

Twenty years ago, the increased occupant productivity linked with high performance, LEED, green, sustainable, Passivhaus, and day-lit buildings was generally backed by anecdote. A flurry of recent research, however, has consistently confirmed the once-anecdotal narrative: when buildings perform better, workers do more, students learn more, and sales spike. After a substantial financial investment in a high performance building, owners can expect meaningful energy savings with modest payback times, but for buildings where occupant performance has a value, both the construction costs and energy savings are a rounding error relative to occupant productivity benefits. Geothermal and passive thermal systems are explored as opportunities to align low-energy thermal systems to acoustics; passive ventilation and thermal mass are explored as low-energy thermal comfort strategies that challenge acoustic concerns. The author's recent research in daylighting, thermal mass, night insulation, and the effects of cafeteria noise on the eating habits of elementary school children will be highlighted.

Contributed Papers

11:00

4aAA7. Consideration of acoustics in leadership in energy and environmental design (LEED) version 4. John J. LoVerde, Samantha Rawlings, and David W. Dong (Veneklasen Assoc., 1711 16th St., Santa Monica, CA 90404, jloverde@veneklasen.com)

Acoustical criteria have been added to the LEED rating systems with each new revision, beginning with Schools and Healthcare. With the recent Version 4 update, an acoustical credit has been added for the majority of New Construction LEED rating systems. There are weaknesses in the acoustical LEED credits, including imbalance between cost and benefit, mitigation beyond industry standards, and implementation of requirements within non-sensitive areas. This paper reviews the purpose of acoustics within a sustainable design system, identifies areas where the current language hits the mark or should be revised, and adds areas for future consideration of acoustical design within a sustainable design rating system.

11:15

4aAA8. Concrete core activation and suspended ceilings: Designing for comfort, energy efficiency, and good acoustics. Martijn Vercammen and Hanneke Peperkamp (Peutz, Lindenlaan 41, Mook 6585 ZH, Netherlands, m.vercammen@peutz.nl)

The trend to design energy-efficient buildings continues. Both legislation as sustainability assessment methods have increased the popularity of thermally activated concrete slabs. It is a way to use low temperature heating and high temperature cooling which makes it very suited for the use in low energy systems. The efficiency of these systems relate to the surface area, often the ceiling area. Exactly that surface was already the domain for the sound absorbing ceiling. So in new buildings with high energy performance due to concrete core activation, the sound absorption is often banned, resulting in very poor acoustics. The use of open, sound absorbing ceilings will have an influence on the thermal capacity of the concrete slabs, However, little is known about this effect. To investigate the effect of open ceilings to both the cooling capacity as the sound absorption, theoretical/empirical models have been made to estimate the effect on the cooling capacity and the sound absorption. The method is also tested in a field situation. It turns out that optimization is possible, with both cooling capacity as sound absorption around 70% of the maximum.

11:30

4aAA9. Acoustics testing and simulation analysis of waiting hall in the line-side high-speed railway station. Gang Liu, Dan Hou, Lixiong Wang, and Rui Dang (School of Architecture, Tianjin Univ., Wei Jin Rd. No. 92, Nankai District, Tianjin 300072, China, youknowleft@sina.com)

Integrated the flow density changes of the waiting hall in a line-side high-speed railway station, an on-site measurement of noise environment is carried out. The characteristics of the acoustic environment are discussed in this study. Furthermore, combining the measured data of the background noise, acoustic computer simulation program ODEON calculates the reverberation time and speech transmission index of public broadcasting system. The results indicate that the reverberation time exceeds 5 s and the speech intelligibility of the south waiting room and dining area in second floor is lower than 0.4. Against the existing problems, various scenarios for upgrading the acoustic environment of waiting room are presented and proved to be efficient. Moreover, from the optimization process, it is certificated that the requirement of speech intelligibility can be also satisfied when appropriately relaxing the reverberation time limits specified in the regulation.

11:45

4aAA10. Acoustical considerations in design and construction of Turkish Contractors Association headquarters. Zühre Sü Gül (R&D, MEZZO Studyo LTD., METU Technopolis KOSGEB-TEKMER No112, ODTU Cankaya, Ankara 06800, Turkey, zuhre@mezzostudyo.com) and Mehmet Caliskan (Mech. Eng., Middle East Tech. Univ., Ankara, Turkey)

A LEED candidate building of Turkish Contractors Association Headquarters is a prestige building located in Ankara, the project and construction of which is sponsored by leading contractor companies of Turkey. As it represents the status of Turkish construction industry, from concept phase to the very recent inauguration of the building, the major consideration has been the application of latest technology and concepts such as sustainability within the scheme of building that can pioneer future works in the field. Chilled beam ventilation is one example of new technologies applied in the building system design that takes into account high energy efficiency with minimum use of fuel or natural sources. In acoustical terms, the building envelope and structural members together with interior and environmental noise sources in relation to the building services are studied. In order to provide acoustical comfort levels in acoustically sensitive spaces and to control noise and vibration at the source and sound paths, materials and methods are developed. Specifically acoustical interventions and solutions proposed for multi-purpose hall, offices, board and meeting rooms, foyers, mechanical rooms, roof-top units, and generators located close by at the site are discussed within the context of this paper.

THURSDAY MORNING, 8 MAY 2014 — 554 A/B, 8:00 A.M. TO 12:00 NOON

Session 4aAB

Animal Bioacoustics: Acoustics as a Tool for Population Structure II

Shannon Rankin, Cochair
Southwest Fisheries Sci. Ctr., 8901 La Jolla Shores Dr., La Jolla, CA 92037

Kathleen Stafford, Cochair
Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105

Contributed Papers

8:00

4aAB1. Detecting and locating manatees in a zero visibility environment. Mario R. Rivera-Chavarria (Comput. Sci., Universidad de Costa Rica, Centro de Investigaciones en Tecnologías de la Información y Comunicación, Universidad de Costa Rica, Sede, Montes de Oca, 30101, San Jose, Costa Rica, San Jose 2060, Costa Rica, mariorrivera@gmail.com), Hector Guzman (Smithsonian Tropical Res. Inst., Panama, Panama), and Jorge Castro (Centro Nacional de Alta Tecnologia, San Jose, Costa Rica)

Manatees are an endangered species, and their population numbers have not been estimated in Panamian wetlands. Traditional monitoring methodologies using aerial surveys and visual sighting are ineffective for such turbid environments. We did a nine-month passive-acoustic survey using a Kayak with a stereo hydrophone array to detect and locate west Indian manatees in San San Pond Sak and Changuinola rivers in Panama. Twice a day transects with a total covering of 1700 km resulted 110 localizations and the recording of 1339 manatee vocalizations. Individual counting and the identification of biologically relevant sites is possible based on passive acoustics. Only a 2% of the acoustic detections were accompanied by clear sightings, indicating that visual census methodologies are ineffective in turbid environments and drastically underestimate manatee populations. We recommend this low-cost methodology to estimate manatee population more reliably in previously unsurveyed areas.

8:15

4aAB2. Fin whale song characteristics recorded on ocean bottom seismometers in the Northeast Pacific Ocean. Michelle Weirathmueller and William S. D. Wilcock (School of Oceanogr., Univ. of Washington, 1503 NE Boat St., Seattle, WA 98105, michw@uw.edu)

Fin whales produce low frequency sequences of vocalizations that can be detected opportunistically on ocean bottom seismometers (OBSs). Using an automatic detection algorithm, we have analyzed fin whale calls recorded on OBSs in the Northeast Pacific Ocean over broad spatial and temporal scales. The Cascadia Initiative experiment consists of 70 OBSs deployed for a total of four years (2011–2015). It extends from Vancouver Island to Cape Mendocino, and several hundred kilometers offshore. Additional OBS data that overlap spatially with the northern portion of the Cascadia Initiative instruments are available from the Neptune Canada cabled observatory, which has been online since 2009, and from standalone deployments between 2003 and 2006. With this study, we examine call characteristics and seasonal call counts for patterns that might indicate migratory movements or distinct acoustic populations. Both frequency and inter-pulse interval (IPI) are automatically extracted for each detected call and seasonal and inter-annual calling patterns are examined using daily binned call count histograms. Preliminary analysis of a subset of Cascadia Initiative data from 2011 to 2013 shows a dominant sequence of alternating classic and backbeat calls at center frequencies of 20 and 18.5 Hz, respectively, and preceding IPIs of 16 and 18 s, respectively.

8:30

4aAB3. Characteristics of sounds detected and localized in Hawaiian waters in Oct. 2013 believed to be from a Bryde's whale. Stephen W. Martin and Brian M. Matsuyama (SSC PAC, 53560 Hull St., Code 71510, San Diego, CA 92152, steve.w.martin@navy.mil)

Pulsed acoustic sounds suspected to be from a single Bryde's whale (*Balaenoptera edeni*) were automatically detected and localized in real time between 1130 and 1304 local time on 6 August 2013 utilizing hydrophones at the Pacific Missile Range Facility, Hawaii. The bottom mounted hydrophones are located 40 km to 80 km northwest of the Napali Coast of Kauai in waters over 4 km in depth. The localized sounds moved from east to west on a course of 294 degrees true for a distance of ~21.6 km with an average speed of 13.8 km/h, which is within the range reported for Bryde's whales. The sounds resemble those previously identified as being from Bryde's whales associated with visual sightings (Oleson *et al.* 2003) and acoustic only observations (Heimlich *et al.* 2005). Detailed analysis of the sounds

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revealed 27 emissions over the period with an inter-pulse interval of 216.6 s (SD, 69.4 s; range, 33–358 s). The duration of the sounds was approximately 1.8 s with major energy apparent at 33 Hz exhibiting burst tonal characteristics often with lower frequency tonal content. Generic calibration data for the hydrophones allows estimation of the source levels of the sounds, which fit within the range previously reported for the species (Cummings *et al.* 1986).

8:45

4aAB4. Trends and variations in the baseline soundscape of America's first offshore wind farm. T Aran Mooney, Maxwell B. Kaplan, Annamaria Izzi, and Laela Sayigh (Biology Dept., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., Woods Hole, MA 02543, amooney@whoi.edu)

With the development of Cape Wind, Nantucket Sound, Massachusetts may become home to America's first offshore wind farm. The goal of this ongoing project is to establish the baseline (pre-construction) soundscape of anthropogenic and biological activity, including diel and seasonal variability of various sound types, at the construction site and nearby comparison sites. Acoustic recorders have been deployed since April 2012, recording on a 10% duty cycle (sample rate: 80 kHz). Multiple fish sounds have been identified with the predominant signals attributed to cusk eels (Family Ophidiidae). Cusk eel sounds consist of a series of pulses, with energy between 400 and 2500 Hz. They are detectable from April to October, with dense choruses occurring during the summer months. Sound energy levels during these choruses increased near the hours of sunrise and sunset. Vessel traffic also showed diel and seasonal trends, with peaks during the daytime and in the summer. These trends in biological and human activity provide key baseline records for evaluating the possible influence of wind farm construction and operation on a local US soundscape.

9:00

4aAB5. Algorithmic analysis of sounds using morphometric methods. Benjamin N. Taft (Landmark Acoust. LLC, 1301 Cleveland Ave., Racine, WI 53405, ben.taft@landmarkacoustics.com)

The vast diversity of animal sounds makes it difficult to analyze them in a quantitative, yet general way. Morphological research faces the same scope of variation and has met the challenge of generally applicable quantitative analysis by using landmarks to describe shapes. Spectrograms are a ubiquitous tool in bioacoustics research because they portray sounds visually—as shapes. The SoundPoints application provides a modular set of algorithms that reduce time-varying signals, especially sounds, into sets of landmarks. The landmark stage of analysis provides a layer of abstraction between feature detection and statistics or pattern recognition algorithms. As a result, it is possible to measure the large numbers of sounds that are needed to quantify variation at individual, population, and species levels. To demonstrate measures of stereotypy, I will present a developmental series of Swamp Sparrow (*Melospiza georgiana*) calls composed of more than 600 000 individual notes. To demonstrate spatial applications of classification, I will use a meta-population analysis of similarity among 22 000 Tree Swallow (*Tachycineta bicolor*) dawn song syllables.

9:15

4aAB6. Acoustic characterization and vocal behavior of North Atlantic right whale surface active groups. Edmund R. Gerstein (Charles E Schmidt College of Sci., Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33486, gerstein2@aol.com), Vasilis Trygonis (FAU / Harbor Branch Oceanogr. Inst., Lesvos island, Greece), Jim Moir (Marine Resources Council, Stuart, FL), and Steve McCulloch (FAU / Harbor Branch Oceanogr. Inst., Fort Pierce, FL)

Focal acoustic surveys were conducted to assess the vocal behavior of North Atlantic right whales in the shallow waters of the southeast critical habitat. Underwater vocalizations were archived using autonomous buoys in close proximity to surface active groups (SAGs) providing sound production data vital for regional passive acoustic monitoring and conservation. Classification trees were used to examine the distinguishing characteristics of calls and quantify their variability within the surface active groups vocal repertoire. Calling rates were higher than those reported in the Bay of Fundy, which may be a factor of habitat demographics. Sound production rate and call type usage were correlated with group cohesion, revealing a consistent call distribution pattern across SAGs of varying sizes and composition. The within-bout clustering probability of low and high frequency calls suggest that temporal affinities between vocalization classes may be indicators of shared social functions. The results demonstrate that concurrent temporal and spectral analysis is powerful for investigating and presenting the interrelationships of calls with social behavior and group composition.

9:30

4aAB7. Using relative Doppler from multiple observations of dolphin whistles as an aid to localization and tracking. Paul Hursky (HLS Res. Inc., 3366 North Torrey Pines Court, Ste. 310, La Jolla, CA 92037, paul.hursky@hlsresearch.com)

A variety of techniques have been used to track marine mammals from their vocalizations. For example, hyperbolic fixing and cross-fixed beams are well-developed approaches, both requiring multiple separated sensors. Dolphin vocalizations, consisting of clicks and whistles, pose some interesting challenges as well as opportunities. Dolphins are often observed in groups and their vocalizations can be quite dense in time. Before their sounds can be used on separated sensors, they must be associated, so that different sounds, for example, are not mistakenly assumed to be the same sound. Since it is often difficult to distinguish different clicks, it becomes difficult to associate (and thus track) them, when there are a lot of them (either from the same animal, or from many animals). By contrast, whistles typically are much easier to associate, even if overlapping. Whistles last seconds at a time and have distinctive melodies that span tens of kilohertz in bandwidth, often with harmonics. The high frequency of dolphin whistles and the fact that these animals are in constant motion suggests a novel feature to incorporate into their tracking—we discuss using relative Doppler, estimated from observations of whistles on multiple separated sensors, as an aid to localization and tracking.

9:45

4aAB8. Large-scale automatic acoustic monitoring of African forest elephants' calls in the terrestrial acoustic recordings. Yu Shiu, Peter H. Wrege, Sara Keen, and Elizabeth D. Rowland (BioAcoust. Res. Program, Cornell Univ., 159 Sapsucker Woods Rd., Ithaca, NY 14850, atoultaro@gmail.com)

African forest elephants live in the rain forests of western and central Africa. The dense habitat prevents them from communicating visually within the family group. Automatic detection of African Forest Elephants' calls intercepts signals in their communication channel and enables fast processing of large scale acoustic data. In this work, first, an automatic detection system targeting at African forest elephants' rumble calls is proposed. De-noising pre-processing, design of acoustic feature vectors, and choice of classifiers are discussed respectively. Second, the detector's performance is evaluated by the cross-validation of a 432-h of acoustic recording from eight locations in Gabon, Africa. It shows that the detector achieve 79.19% true positive rate when the false positive number is the low 5.70 per hour. The F1-score (geometric mean of precision and recall) is around 0.77 when relatively high score threshold (over 0.8) is selected. Finally, a case study demonstrates the results of applying our automatic detection system to a large-scale data set, which amounts to 420 days of acoustic recording over 3 years from the Ivindo National Park, Gabon. Visualization of the call activities reveal the seasonal and daily patterns as well as the temporal variation over the 3 years.

10:00

4aAB9. Single-sensor, cue-counting density estimation of highly broadband marine mammal calls. Elizabeth T. Küsel, Martin Siderius (Dept. of Elec. and Comput. Eng., Portland State Univ., Ste. 160, 1900 SW 4th Ave., Portland, OR 97201, ekusel@pdx.edu), and David K. Mellinger (Oregon State Univ. and NOAA Pacific Marine Environ. Lab., Newport, OR)

Odontocete echolocation clicks have been used as a preferred cue for density estimation studies from single-sensor data sets, studies that require estimating detection probability as a function of range. Many such clicks can be very broadband in nature, with 10-dB bandwidths of 20 to 40 kHz or more. Because detection distances are not realizable from single-sensor

data, the detection probability is estimated in a Monte Carlo simulation using the sonar equation along with transmission loss calculations to estimate the received signal-to-noise ratio of tens of thousands of click realizations. Continuous-wave (CW) analysis, that is, single-frequency analysis, is inherent to basic forms of the passive sonar equation. Considering transmission loss by using CW analysis with the click's center frequency while disregarding its bandwidth has recently been shown to introduce bias to detection probabilities and hence to population estimates. In this study, false killer whale (*Pseudorca crassidens*) clicks recorded off the Kona coast of Hawai'i are used to quantify the bias in sonar equation density estimates caused by the center-frequency approach. A different approach to analyze data sets with highly broadband calls and to correctly model such signals is also presented and evaluated. [Work supported by ONR.]

10:15–10:30 Break

10:30

4aAB10. Residency of reef fish during pile driving within a shallow pierside environment. Joseph Iafrate, Stephanie Watwood (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, joseph.iafrate@navy.mil), Eric Reyier (InoMedic Health Applications, Inc., Environ. Services, Kennedy Space Ctr. Ecological Program, FL), Matthew Gilchrest, and Steven Crocker (McLaughlin Res. Corp., Naval Undersea Warfare Ctr., Newport, RI)

The potential effects of pile-driving on fish populations have received significant attention with the prevalence of construction at in-shore areas throughout the world. In this study, the movement and survival of free-ranging reef fish in Port Canaveral, Florida, in response to pile driving for 35 days at an existing wharf was examined through the use of acoustic telemetry. Twenty-seven Sheepshead (*Archosargus probatocephalus*) and 13 mangrove snapper (*Lutjanus griseus*) were monitored for a period of approximately 11 months. Underwater acoustic receivers were deployed within Port Canaveral to complement an existing array of compatible receivers spanning a range of over 300 kilometers (km) along the east coast of Florida. Baseline residency and diel patterns of movement were compared for fish in two adjacent locations with and without disturbance before, during, and after the event. There was a significant decline in residency index for mangrove snapper at the construction wharf noted during the pre-during period. Also, 16 of 25 fish tagged at the construction wharf were detected 3-months post tagging, and 11 fish were detected 6-months post tagging. Although there was no apparent impact on patterns of behavior for resident reef fish populations, alterations on behavior of individual fish were noted, including displacement.

10:45

4aAB11. Investigating the relationship between foraging odontocetes and ocean acoustic biomass off the Kona coast of the Island of Hawaii. Adrienne M. Copeland (Hawaii Inst. of Marine Biology, Univ. of Hawaii, P.O. Box 1106, Kailua, HI 96734, acopelan@hawaii.edu), Whitlow Au, Giacomo Giorli (Hawaii Inst. of Marine Biology, Univ. of Hawaii, Honolulu, HI), and Jeffrey Polovina (Pacific Islands Fisheries Sci. Ctr., NOAA, Honolulu, HI)

To understand the distribution of deep diving odontocetes, it is important to investigate the relationship between foraging whales and their prey. Tagged sperm whales have been documented to dive as deep as 1202 m. Short-finned pilot whales in Hawaii dive deeper during the day down to 600–800 m and shallower dives at night, driven possibly by the migration of organisms at night. Foraging sperm and pilot whales off the Island of Hawaii were located using a hydrophone array detecting echolocation clicks. A 500 m by 500 m active acoustics survey box was set up over two foraging sites: one during the night above foraging sperm whales and one during the day over foraging pilot whales. A four-frequency (38, 70, 120, and 200 kHz) split-beam echosounder collected acoustic data over foraging populations and non-foraging control sites of a similar bottom depth and time. The Nautical Acoustic Scattering Coefficient (NASC) or acoustic biomass ($m^2 nmi{-}2$) profile over the complete water column was statically compared over foraging and non-foraging populations to analyze the relationship between foraging and ocean biomass.

11:00

4aAB12. Estimating the range of Baleen whale calls recorded by hydrophone streamers during seismic surveys. Shima H. Abadi (Lamont–Doherty Earth Observatory, Columbia Univ., 122 Marine Sci. Bldg., University of Washington 1501 NE Boat St., Seattle, Washington 98195, shimah@ldeo.columbia.edu), Timothy J. Crone, Maya Tolstoy (Lamont–Doherty Earth Observatory, Columbia Univ., Palisades, NY), William S. D. Wilcock (School of Oceanogr., Univ. of Washington, Seattle, WA), and Suzanne M. Carbotte (Lamont–Doherty Earth Observatory, Columbia Univ., Palisades, NY)

Like all sources of anthropogenic sound in the oceans, seismic surveys have the potential to disturb marine mammals and impede their communications. Since airgun arrays produce a considerable amount of low frequency energy, their impact on Baleen whales may be most significant. For these reasons, extensive mitigation efforts accompany seismic surveys, including visual and acoustic monitoring, but additional approaches could be useful to verify these efforts and study the behavior of whales. One approach is to utilize the hydrophone streamer to detect and locate calling Baleen whales. To develop this method, data are being analyzed from a seismic reflection survey conducted with the R/V Langseth off the coast of Washington in summer 2012. The seismic streamer is 8 km long with 636 hydrophones sampled nearly continuously at 500 Hz. The work focuses on time intervals when only a mitigation gun is firing because of marine animal sightings or turns at the ends of lines. Ranges and orientations are estimated by calculating the signal arrival angles for different groups of receivers. Data from the marine mammal observers on the R/V Langseth and other ships in the area are used to verify the analysis. [Sponsored by NSF.]

11:15

4aAB13. Soundscapes and vocal behavior of humpback whales in Massachusetts Bay. Nathan D. Merchant, Susan E. Parks (Dept. of Biology, Syracuse Univ., 107 College Pl., Syracuse, NY 13244, ndmercha@syr.edu), Sofie M. Van Parijs (Northeast Fisheries Sci. Ctr., NOAA Fisheries, Woods Hole, MA), David N. Wiley, Michael A. Thompson (Stellwagen Bank National Marine Sanctuary, Scituate, MA), and Ari S. Friedlaender (Duke Univ. Marine Lab., Beaufort, NC)

In recent years, technological advances have revolutionized the study of acoustic communication in marine mammals. Exciting new perspectives on vocal behavior, acoustic habitats, and the influence of noise on communication are offered by passive acoustic monitoring (PAM) platforms such as acoustic tags (DTAGs), autonomous PAM recorders, drifting PAM buoys, and subsea gliders. These innovations bring the opportunity to integrate data from fixed and mobile PAM devices to gain deeper insight into the dynamic interactions between marine mammal vocalizations, behavioral context, and the acoustic environment. In this study, we bring together such data sources to study the vocal behavior and acoustic habitat of humpback whales in the context of their spring and summer feeding grounds. Recordings were made in Stellwagen Bank National Marine Sanctuary during 2008–2010, using arrays of autonomous PAM recorders and DTAGs. In addition, AIS ship-tracking data were obtained to study the influence of vessel movements. We present preliminary findings of this work and discuss future strategies for analyzing the spatiotemporal interactions between vocal behavior and acoustical context.

11:30

4aAB14. Measuring the sonic, infrasonic and seismic soundscape of the Southern White Rhinoceros *(Ceratotherium simum simum)* at a wildlife park conservation center. Suzi Wiseman (Environ. Geography, Texas State Univ.-San Marcos, 3901 North 30th St., Waco, TX 76708, sw1210txstate@gmail.com), Preston S. Wilson (Mech. Eng., Univ. Texas at Austin, Austin, TX), and Frank Sepulveda (Geophys., Baylor Univ., Killeen, TX)

Many creatures, including the myopic rhinoceros, depend upon hearing and smell to determine their environment. Nature is dominated by biophonic and geophonic sounds quickly absorbed by soil and vegetation, while anthrophonic urban soundscapes exhibit vastly different physical and semantic characteristics, reflecting off hard geometric surfaces, distorting and reverberating, and becoming noise. Noise damages human

physiologically, including reproductively, and likely damages other mammals. Rhinos vocalize sonically and infrasonically but audiograms are unavailable. They generally breed poorly in urban zoos, where infrasonic noise tends to be chronic. Biological and social factors have been studied but little attention if any has been paid to soundscape. To comprehensively describe the rhinos' sonic, infrasonic and seismic environment at Fossil Rim Wildlife Center, one of the few U.S. facilities to successfully breed white rhinos in recent years, I began by comparing the sound metrics at different times of day in categories, for example, during visitation hours versus park closure. Further analysis will seek particular parameters known to be injurious to humans, plus those already known to impact animals. Later, the soundscapes of other facilities could be compared to seek correlations between their soundscapes and the health and well-being of the rhinos within their care.

11:45

4aAB15. Near real-time detection, beam-forming, and telemetry of marine mammal acoustic data on a wave glider autonomous vehicle. Harold A. Cheyne, Dean Hawthorne (Lab of Ornithology, Cornell Univ., 95 Brown Rd., Rm. 201, Ithaca, NY 14850, haroldcheyne@gmail.com), Charles R. Key, and Michael J. Satter (Leidos, Long Beach, MS)

Impacts of anthropogenic noise on marine mammals are becoming increasingly important for regulatory and research study, yet assessing and mitigating these impacts is hindered by current technology: archival underwater acoustic recorders have their data analyzed months after the activity of interest, and towed hydrophone arrays suffer from nearby ship and seismic air gun noise. This work addresses these drawbacks by developing an acoustic data acquisition and transmission system for use with a Wave Glider, to provide near real-time data for marine mammal monitoring and mitigation. The goal of the system is to be capable of months of autonomous monitoring in areas that would otherwise not be surveyed, and to transmit acoustic data within minutes of acquisition to enable rapid mitigation. Sea tests have demonstrated the proof-of-concept with the system recording four channels of acoustic data and transmitting portions of those data via satellite. Ongoing work is integrating a detection-classification algorithm onboard the Wave Glider and a beam-forming algorithm in the shore-side user interface, to provide the user with a topographic view of the Wave Glider; a sound source direction estimate; and aural and visual review of the detected sounds.

THURSDAY MORNING, 8 MAY 2014 — BALLROOM E, 9:00 A.M. TO 11:20 A.M.

Session 4aBA

Biomedical Acoustics: Biomedical Applications of Low Intensity Ultrasound I

Thomas L. Szabo, Chair
Biomedical Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215

Invited Papers

9:00

4aBA1. Low intensity ultrasound—Diverse biomedical applications. Thomas L. Szabo (Biomedical Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215, tlszabo@bu.edu)

Low intensity ultrasound has been useful in a surprisingly wide range of biomedical applications. Ultrasound can affect both the central nervous system (CNS; brain and spinal cord) and the peripheral nervous system (PNS). Ultrasound methods offer the possibility of stimulating receptor structures and unmyelinated nerve fibers not just on the surface but also those otherwise inaccessible, deeper in the body or brain. Preliminary work indicates that information can be transferred via unmyelinated nerve fibers when normal avenues of sensing are damaged or inoperative. Tones and acoustic speech have been directly transferred through nerves without direct hearing. Radiation force effects can provide a variety of sensational effects and can be employed in ultrasound dynamic tactile arrays. Early prototype devices for diagnosis of diseases of hearing and audio prosthetics were developed in the early 1980s in the Soviet Union by Gavrilov and Tsirulnikov. Neuromodulation of the brain transcranially has demonstrated activation of motor responses. Low intensity ultrasound has also been applied to bone and wound healing. In a related application, it has been applied to the growth of artificial neovessels.

9:20

4aBA2. Ultrasonic neuromodulation: Three conjectures common to the peripheral and the central nervous systems. Robert Muratore (Quantum Now LLC, 49 Cedar Dr., Huntington, NY 11743, wave@quantumnow.com)

The nervous system responds with finesse to incident ultrasound. Three conjectures are proposed, abstracted from the literature, which illustrate commonalities in the response of the peripheral and the central nervous systems, and serve as an introduction to the nascent field of ultrasonic neuromodulation. (1) A mathematical function fit to neuronal effect vs. acoustic dose has a root at a non-trivial dose. Above a threshold dose, nerves and brain regions are stimulated; at very high doses, normal neuronal activity is inhibited. Thus, there exists an intermediate dose balancing stimulus and inhibition. (2) An acoustic beam can modulate a neuronal region larger than

that which it insonifies. Nerves can be stimulated by insonifying a small portion of their axon. Brain regions exhibit responses, such as spreading depression, to localized insonification. (3) The spatial precision of ultrasonic neuromodulation can be considerably finer than the incident acoustic beam width. Thicker nerve fibers are more resistant to the effects of incident ultrasound than are thinner fibers in the same nerve. Across the cortex, displacements of acoustic beams smaller than the beam width can achieve fine motor control. Each of these conjectures plays a role in current neuromodulation experiments.

9:40

4aBA3. Capacitive micromachined ultrasonic transducers with integrated electronics for neuromodulation applications. Butrus T. Khuri-Yakub (E. L. Ginzton Lab., Stanford Univ., Spilker 217, Stanford, CA 94305, khuri-yakub@stanford.edu)

Capacitive Micromachined Ultrasonic Transducers (CMUT) are being made in practically any size (microns to mms), shape (flat or curved), and type (single element, 1-D array, 2-D array, rings, and annular arrays), and at frequencies from 10s of kHz to almost 100 MHz. Along with the transducers themselves, front-end electronics are being integrated as well to provide better performance and enable the use of arrays with a very large number of elements. One important aspect of these integrated arrays is that they can be used for imaging (anatomic and photo-acoustic functional), therapy (high intensity focused ultrasound), and more recently neuro-modulation. This talk will review CMUTs and the methods of integration, then show examples of ultrasound stimulation of lipid bilayers and Salamander retina. We show that the retina responds to ultrasound stimulation as well as it responds to light stimulation and that when the retina's optical response is suppressed chemically it still responds to ultrasonic stimulus. We postulate the possibility of using CMUT 2D arrays as contact lens prosthetic devices capable of restoring some vision in some type of blindness.

10:00

4aBA4. Localization of ultrasound induced *in-vivo* neurostimulation in the mouse model. Randy L. King (DSFM, US FDA, WO62 rm 2217, 10903 New Hampshire Ave., Silver Spring, MD 20993-0002, Randy.King@fda.hhs.gov)

Developments in the use of ultrasound to stimulate and modulate neural activity have raised the possibility of using ultrasound as a new investigative and therapeutic tool in brain research. The phenomenon of ultrasound induced neurostimulation has a long history dating back many decades, but until now there has been little evidence demonstrating a clearly localized effect in the brain, a necessary requirement for the technique to become genuinely useful. Here, we report clearly distinguishable effects in sonicating rostral and caudal regions of the mouse motor cortex. Motor responses measured by normalized EMG in the neck and tail regions changed significantly when sonicating the two different areas of motor cortex. Response latencies varied significantly according to sonication location suggesting that different neural circuits are activated depending on the precise focus of the ultrasound beam. Taken together our findings present good evidence for being able to target selective parts of the motor cortex with ultrasound neurostimulation in the mouse, an advance that should help to set the stage for developing new applications in larger animal models including humans.

10:20

4aBA5. Ultrasound for microvascular tissue engineering. Diane Dalecki (Biomedical Eng., Univ. of Rochester, 310 Goergen Hall, Rochester, NY 14627, dalecki@bme.rochester.edu) and Denise C. Hocking (Pharmacology and Physiol., Univ. of Rochester, Rochester, NY)

A critical obstacle currently facing the field of tissue engineering is the need for rapid and effective tissue vascularization strategies, both during construct development and upon implantation. To address this challenge, we have developed an ultrasound technology for microvascular tissue engineering. The technology utilizes radiation forces in an ultrasound standing wave field to rapidly and non-invasively spatially pattern cells in 3D within hydrogels. Ultrasound-induced patterning of endothelial cells accelerates the emergence of capillary-like sprouts stimulates cell-mediated collagen fibril alignment and results in the maturation of sprouts into lumen-containing microvessel networks throughout collagen hydrogels. Importantly, the morphology of resultant microvessel networks can be controlled by design of the acoustic field employed during fabrication. Specifically, the technology can produce microvascular networks having two distinct, physiologically relevant morphologies; one composed of a tortuous, capillary-like network, and one composed of hierarchical branching vessels (arteriole/venule-like). We have extended the versatility of the technology to lymph endothelial cells and have demonstrated the ability to engineer 3D lymphatic microvessel structures. Thus, this ultrasound technology holds promise as a new approach to induce microvascular network formation and direct vascular morphology in engineered tissues.

10:40

4aBA6. Low intensity (55 kPa) 20 kHz ultrasound heals venous ulcers. Joshua Samuels (Dept. of Biomedical Eng., Drexel Univ., Philadelphia, PA), Michael S. Weingarten (Dept. of Surgery, Drexel Univ. College of Medicine, Philadelphia, PA), Leonid Zubkov, Christopher Bawiec, Youhan Sunny (Dept. of Biomedical Eng., Drexel Univ., Philadelphia, PA), Jane McDaniel, Lori Jenkins (Dept. of Surgery, Drexel Univ. College of Medicine, Philadelphia, PA), David Margolis (Dept. of Epidemiology, Univ. of Pennsylvania Perelman School of Medicine, Philadelphia, PA), and Peter Lewin (Dept. of Biomedical Eng., Drexel Univ., 3141 Chestnut St., BIOMED DEPT, Philadelphia, PA 19104, plewin@coe.drexel.edu)

We report the results of a second clinical pilot study ($n = 19$) involving treatment of chronic wounds (venous ulcers) using novel, fully wearable ultrasound array applicator operating at 20 kHz and generating pressure amplitudes close to 55 kPa (about 100 mW/cm^2, Sptp). The applicator was designed as compact, tether-free, device that can be comfortably worn by subjects at home, permitting active (combined with traditional compression) therapy away from the clinical setting. Patients with venous ulcers documented for over 8 weeks were enrolled from the Drexel Wound Healing Center and, following consent, were randomly assigned into treatment or control groups. Patients were treated weekly (15 min) for a maximum of 12 visits or until wound closure. Treatments were in addition to standard of care compression therapy as ordered by the physician. Of the patients receiving at least three treatments ($n = 16$), the ultrasound treated group had statistically improved ($p < 0.04$) rate of wound closure (reduction of 8.2%/wk) compared to the rate of wound closure for the control group (increase of 7.5%/wk on average). This study represents further proof of the potential healing power of low intensity, low frequency ultrasound. Optical measurements and *in-vitro* work continue to support these findings as well.

11:00

4aBA7. Enhanced fracture repair and mitigation of fracture-healing risk factors using low-intensity pulsed ultrasound. Christopher R. Brodie (Bioventus LLC, 4721 Emperor Blvd, Ste. 100, Durham, NC 27703, chris.brodie@bioventusglobal.com) and Andrew Harrison (Bioventus LLC, York, United Kingdom)

Low-intensity pulsed ultrasound (LIPUS) is used clinically to enhance fracture healing. Level-I clinical studies demonstrate that a specific signal (1.5 MHz ultrasound pulsed at 1 kHz, 20% duty cycle, 30 mW/cm^2 SATA) can accelerate the healing of acute fractures. This result remains a unique benefit of LIPUS, and to date, no other drug or device has been approved by the FDA for accelerated fracture repair. The same signal has been shown in many studies to heal a high proportion of non-union fractures. LIPUS appears to be effective for all three types of non-unions—atrophic, oligotrophic and hypertrophic—even in the absence of revision surgery. The findings are broadly applicable to orthopedics, with similar results regardless of fracture type, fracture location and fracture-management technique. Given the varied causes of non-union, the ability of LIPUS to overcome a high proportion of obstacles to healing indicates that the signal is likely to have pleiotropic effects on multiple cell types within the healing process. Smoking, age, and diabetes are known risk factors for delayed union and nonunion. Clinical data, including randomized controlled trials and a registry of 1546 nonunion patients, suggest that LIPUS mitigates these risks and restores the course of normal bone healing.

THURSDAY MORNING, 8 MAY 2014 — 550 A/B, 8:15 A.M. TO 12:00 NOON

Session 4aEA

Engineering Acoustics: Session in Honor of Stanley Ehrlich

David A. Brown, Cochair
ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723

Kenneth G. Foote, Cochair
Woods Hole Oceanogr. Inst., 98 Water St., Woods Hole, MA 02543

Chair's Introduction—8:15

Invited Papers

8:20

4aEA1. Modal transducers and Stan Ehrlich. John L. Butler (Eng., Image Acoust., Inc, 97 Elm St., Cohasset, MA 02025, jbutler@imageacoustics.com)

There have been a number of transducer designs which use the dipole mode of a piezoelectric cylinder to obtain directionality since Stan Ehrlich's early patent (with P.D. Frelich), ["Sonar Transducer," U.S. Patent 3,290,646, December 6, 1966] was first published. There is now a whole class of transducers, called vector sensors or hydrophones, which use the dipole mode in one or more directions. In addition to this, other designs have emerged which use modes higher than the monopole and dipole modes. For example, the added use of the quadrupole mode has allowed beam patterns from cylinders which approximate patterns from piston transducers. Work on the dipole mode and higher modes of spherical transducers and arrays allow 3-D acoustical coverage from one transducer or array. This presenters interest in modal excitation from transducers and arrays began after reading Stan's patent, and interest developed further after working with Stan at Raytheon. A review of some of the transducers and arrays which we worked on will be presented, along with more recent work based on these modal concepts.

8:40

4aEA2. Multimode and other sonar transducer patents of Stanley Ehrlich. David A. Brown (Elec. Engineering/ATMC, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

Stanley Ehrlich was an innovative sonar transducer designer and inventor who made many important contributions to the field of acoustic transduction at Raytheon Submarine Signal Division in Portsmouth, Rhode Island. With the advent of searchable digital patent archives at the USPTO and google/patents, it is now relatively easy to review Stan's patents and gain a glimpse at his innovation and creativity. This presentation reviews some these patents including: Sonar Transducer, that describes a multimode transducer producing simultaneously two dipole patterns with mutually perpendicular acoustic axes and an omnidirectional pattern; a Spherical [Multimode] Acoustic Transducer (#3732535 filed 1969) that enables the radial and circumferential vibrating modes of the acoustically excited sphere to be processed to determine bearing. The presentation also draws connections of these and other Ehrlich inventions to more recent ongoing works in multimode transducers.

9:00

4aEA3. Nearfield of an electroacoustic transducer, with implications for performance measurement. Kenneth G. Foote (Woods Hole Oceanogr. Inst., 98 Water St., Woods Hole, MA 02543, kfoote@whoi.edu)

The spatial structure of the nearfield of an electroacoustic transducer is known to be complicated. This is illustrated by numerical modeling of the nearfield of an ideal planar circular piston in a rigid, infinite baffle. There are implications for performance measurements of electroacoustic transducers including hydrophones in tanks.

9:20

4aEA4. Analysis of nonuniform circular flexural piezoelectric plate transducers. Boris Aronov (ElectroAcoust. Res. Lab. - ATMC, BTech Acoust. LLC, Fall River, MA) and David A. Brown (ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

An analytical treatment of the circular flexural plate transducer having a nonuniform electromechanically active-passive (bilaminar) mechanical system is presented. The analysis is made using the energy method that was previously applied to calculating the parameters of uniform fully active (bimorph) circular plate transducers [Aronov, J. Acoust. Soc. Am. **118**(2), 627–637 (2005)]. It is shown to first order approximation, that the vibration mode shapes do not change significantly for a large range of relative dimensions of the active and passive laminates of the mechanical system considered in terms of optimizing the effective coupling coefficient of the transducer. Therefore, the transducer can be considered as having a single degree of freedom, and its operational characteristics can be calculated using the same technique as previously used for uniform plates. The dependence of the resonance frequencies, effective coupling coefficients, and parameters of the equivalent electromechanical circuit on the relative dimensions of the active and passive laminates for several combinations of the active and passive materials are presented. The main results are in a good agreement with experimental data.

9:40–10:00 Break

Contributed Papers

10:00

4aEA5. Planar microphone based on piezoelectric electrospun poly (γ-benzyl-?,L-glutamate) nanofibers. James E. West, Kailiang Ren (ECE, Johns Hopkins Univ., 3400 N. Charles St., Barton Hall 105, Baltimore, MD 21218, jimwest@jhu.edu), and Michael Yu (ECE, Johns Hopkins Univ., Salt Lake City, Utah)

Velocity and pressure microphones comprised of piezoelectric poly (γ-benzyl, L-glutamate) (PBLG) nanofibers were produced by adhering a single layer of PBLG film to a Mylar diaphragm. The device exhibited a sensitivity of 65 dB/Pa in air, and both pressure and velocity response showed a broad frequency response, which was primarily controlled by the stiffness of the supporting diaphragm. The pressure microphone response was 3 dB between 200 Hz and 4 kHz when measured in a semi-anechoic chamber. Thermal stability, easy fabrication, and simple design make this single element transducer ideal for various applications including those for underwater and high temperature use.

10:15

4aEA6. Experimental results of motional current velocity control intended for broadband piezoelectric projectors. Robert C. Randall (Raytheon, 188 Hanover St. Apt. 3, Fall River, MA 02720, bobrandall81@gmail.com), David A. Brown (Univ. of Massachusetts Dartmouth, Barrington, Rhode Island), and Corey Bachand (BTech Acoust. LLC, New Bedford, MA)

Velocity control with active feedback can be useful for flattening a projector's frequency response, reducing distortion, and mitigating array interaction effects. This has been demonstrated and commercialized for HiFi audio, but has seen little attention for underwater SONAR and communications applications. The benefits and tradeoffs of using velocity control to drive an underwater piezoelectric transducer or array of transducers is presented, comparing array beam patterns both with and without velocity control. The theoretical effectiveness of motional current velocity control is discussed for various piezoelectric loads with coupling coefficients ranging from 0.3 to 0.9. The utility of using a digital feedback amplifier and in situ calibration methods with this approach is discussed. A prototype Class D amplifier using motional current feedback driving an equivalent circuit load for a BTech Acoustics single crystal segmented cylinder is presented. Experimental results of frequency response, bandwidth, and feedback stability are also considered.

10:30

4aEA7. Automated parameter fitting of two-port network transducer models. Daniel M. Warren (Knowles Electronics, 1151 Maplewood Dr., Itasca, IL 60134, daniel.warren@knowles.com)

Networks of linear lumped-parameter components, as may be simulated in electronic circuit design software such as SPICE, are a lightweight, fast, and convenient means of predicting the behavior of electroacoustic transducers in practical application. The development of these models can be quite burdensome. The selection and networking of the equivalent electronic components requires specialized domain knowledge of the transduction mechanisms and design of a transducer. Previously, a general network synthesis approach was proposed [Warren, Daniel, "Applications of network synthesis and zero-pole analysis in transducer modeling," J. Acoust. Soc. Am. **133**, 3360–3360 (2013)] but was deemed to be an unreliable and awkward means of developing transducer networks in practice. However, networks that represent transducer behavior are generally well-known for a given transducer type and design. The more difficult task, or, at least, the more often performed and thereby repetitive task, is the selection of component parameter values which correctly predict the transducer's behavior under all electrical drive and acoustical loading conditions which may be encountered in practical application. The approach taken here is to assume that the network itself is already known and seek to develop an automated means of determining the correct parameter values.

10:45

4aEA8. Model for the design of a pressure actuated self-sustained oscillator as an acoustic projector. Stephen C. Thompson (Graduate Program in Acoust., The Penn State Univ., N-249 Millennium Sci. Complex, University Park, PA 16803, sct12@psu.edu)

Wind musical instruments are examples of a pressure operated self-sustained oscillator that acts as an acoustic projector. Recent studies have shown that this device can also be implemented underwater. However the design parameters for such a device are necessarily different due to the large difference in medium density and acoustic impedance. This paper describes a model that is sufficient to predict the performance of the projector and to understand the effects of design changes on the performance.

11:00–12:00 Panel Discussion

Session 4aID

Interdisciplinary, Public Relations Committee, and Education in Acoustics: Effective Communication Between Acoustics Professionals and the Media

Andrew A. Piacsek, Cochair
Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926

Steven L. Garrett, Cochair
Grad. Prog. in Acoustics, Penn State, Appl. Res. Lab., P. O. Box 30, State College, PA 16804

Chair's Introduction—10:00

Invited Papers

10:05

4aID1. What to expect when you're expecting media calls. Jason S. Bardi (NMS, AIP, 1 Phys. Ellipse, College Park, MD 20740, jbardi@aip.org)

A decade's worth of college, grad school, and post-doc work, countless sleepless nights toiling in your own laboratory, a long route to discovery, your ultimate breakthrough and it has come to this: the phone is ringing. A reporter is on the line. What does she/he want? What should you say? I'm here to tell you, "Don't panic!" You have been preparing for this interview your entire professional career. You are one of the world's leading experts in your area, and that's one of the reasons why the reporter is calling. You also have a story to tell, the reporter wants to hear it, and the interview should be more conversation than inquisition. This talk will help you realize that, helping you make the most of your time in the spotlight by putting the PR and press process into perspective, offering some tips of the trade, describing your rights and responsibilities as a source, and sharing best practices for handling media inquiries.

10:20

4aID2. Why should a U.S. Navy researcher discuss cicada mating calls for hours with several journalist? Derke Hughes (NUWC-DIVNPT, 1176 Howell St., Newport, RI 02841, derke.hughes@navy.mil)

The first journalist to comment officially on my cicada research was by American Institute of Physics (AIP), which was quickly followed up by a LA Times reporter. Unfortunately, the LA Times author wrote how the cicada sound system functioned like a Helmholtz resonator. However, when the author actually interviewed me, I contradicted the journalist by saying that I do not believe that theory was correct. Furthermore, I was interviewed by a radio commentator for National Public Radio (NPR) as well who diligently discussed my research and its inspiration for 50 min. My segment aired for 2 min and those few seconds consisted of "crude construction worker harassment and bar talk." Overall, the opportunity of communicating my research to an international audience was outstanding. The other correspondents that interviewed me for the article were from the *ScienceNow* and *Wall Street Journal*. Also, one of my interviewers was a writer for Le Presse so the immediate columnist coverage did span at least two countries. I am proud to have bought awareness to an insect that has been on earth with civilized mankind for millennia; nonetheless, our practical knowledge of the cicada is rather limited.

10:35

4aID3. Communicating with the media: From the laboratory to the real world. Diana Deutsch (Univ. of California, San Diego, 9500 Gilman Dr. #0109, La Jolla, CA 92037, ddeutsch@ucsd.edu)

Scientists often view communicating with the media as a risky process, based largely on concerns that they might be held responsible for inaccuracies in reporting their work. Yet my experiences with the media have generally been very rewarding. Most frequently, those who have interviewed me have been well prepared and have thought broadly about the subject matter of my research. Our conversations have often induced me to think outside the box and have led to novel ideas for studies that might otherwise have been left undone. The potential for feedback has recently been enhanced by the development of social networks—these often pick up on reports in newspapers and magazines, and provide an important additional forum for discussion. In this talk, I describe some experiences that illustrate these points and offer some suggestions for interacting with the media so as to communicate research findings and their implications most effectively.

10:50

4aID4. We don't bite; we want to get it right. Really. Peter Spotts (The Christian Sci. Monitor, 210 Massachusetts Ave., Boston, MA 02115, pspotts@alum.mit.edu)

If a full-time science writer calls you for an interview, count yourself lucky. These days, full-time science writers are a vanishing breed. The reporter about to interview you may be just as nervous about the impending conversation as you are. You know the subject cold. He or she may have had little time to prepare. We'll take a brief look inside one news organization's day (mine) to understand the context on our side of the so-called divide, and share some thoughts on how you can help us explain what you do to your Aunt Elsie or Uncle Sid.

11:05

4aID5. On becoming an expert witness in a high-profile patent-dispute case. Mark F. Hamilton (Dept. of Mech. Eng., Univ. of Texas at Austin, 204 E. Dean Keeton St., Stop C2200, Austin, TX 78712-1591, hamilton@mail.utexas.edu)

Several years ago, the author was contacted by the legal team representing a major smartphone manufacturer and asked if he would serve as an expert witness in a patent-dispute case to be tried before an administrative law judge at the International Trade Commission. The author had no significant prior experience as an expert witness, and he therefore had no inkling of what responsibilities lay ahead of him. The author will describe his experiences in this case, beginning with assisting the legal team with understanding the relevant acoustics, then writing expert reports, and finally preparing for deposition and trial.

11:20

4aID6. Interviews with the interviewers and interviewees. Andrew A. Piacsek (Phys., Central Washington Univ., 400 E. University Way, Ellensburg, WA 98926, piacsek@cwu.edu)

The basic conflict between scientist and journalist is that each wants to tell a story—but not necessarily the same story. Can they agree on a version that grabs and holds the attention of most people but is still true to the science? Despite the risks of being portrayed inaccurately, should researchers make an effort to talk to the press? This talk will address these questions by synthesizing a series of interviews conducted with two acoustics professionals who have had significant media exposure, a print journalist, a radio journalist, and an academic specializing in science journalism.

11:35–12:00 Panel Discussion

THURSDAY MORNING, 8 MAY 2014 — 557, 8:30 A.M. TO 11:45 A.M.

Session 4aNS

Noise and ASA Committee on Standards: Community Noise

Robert D. Hellweg, Cochair
Hellweg Associates, Wellesley, MA

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

Chair's Introduction—8:30

Invited Papers

8:35

4aNS1. Progress report—American National Standards Institute Community Noise Standard. Bennett M. Brooks (Brooks Acoust. Corp., 30 Lafayette Square - Ste. 103, Vernon, CT 06066, bbrooks@brooks-acoustics.com) and Lawrence S. Finegold (Finegold & So, Consultants, Dayton, OH)

The American National Standards Institute (ANSI) Accredited Standards Committee S12 (Noise) Working Group (WG) 41 has been developing a draft community noise standard document for over 13 years. The purpose of the document is to provide guidance to government officials, acoustical consultants, and other interested persons on how to develop a community noise ordinance or regulation, which is appropriate for the existing local circumstances. The current version of the document embodies significant revisions, based on the inputs of many stakeholders in the community noise arena, including industry, government, consulting, and the public. The document addresses issues such as public and government priorities and values, and available resources, and also provides the technical basis to manage the local sound environment. The keys to the effectiveness of the document are that it provides a menu of options for the user, discusses the trade-offs involved for decisions that must be made by government officials, and emphasizes that enforcement of a community noise ordinance is crucial to its success. Recent progress made by the Working Group in drafting this standard is reported.

8:55

4aNS2. Massachusetts Wind and Noise Technical Advisory Group—Status report. Christopher W. Menge (Harris Miller Miller & Hanson Inc., 77 South Bedford St., Burlington, MA 01776, cmenge@hmmh.com) and Robert D. ONeal (Epsilon Assoc., Inc., Maynard, MA)

In June 2013, the Commonwealth of Massachusetts launched a Community Wind Energy Initiative to provide support and guidance to municipalities, developers and stakeholders for land-based wind projects. The initiative convened a technical advisory group of experts to solicit input on wind turbine sound policy. This Wind and Noise Technical Advisory Group (WNTAG) is led by the Massachusetts Department of Environmental Protection (MassDEP) and includes other state agency representatives, wind energy experts, industry representatives, affected community representatives, health experts, and acoustical consultants. The WNTAG has met several times since July 2013 and has addressed many aspects of wind turbine noise that may influence and/or become a part of a new statewide noise policy for land-based wind turbines. In this presentation, the authors provide perspective on the process, progress toward a revised policy, and the policy and technical aspects that were discussed, which included absolute vs. relative noise criteria, noise level metrics, measurement protocols for compliance evaluation, amplitude modulation, and modeling approaches for pre-construction permitting.

9:15

4aNS3. Regulatory inertia and community noise. Leslie D. Blomberg (Noise Pollution Clearinghouse, Box 1137, Montpelier, VT 05601-1147, les@nonoise.org)

Thirty and forty year old regulations are determining much of the acoustic environment. This paper examines regulations that have not kept up with the times or technology. Aviation, motorcycle, and truck regulations are examined in their historical context, as well as OSHA's backup beeper regulations. Also, in many communities, local noise regulations are decades behind today's best practices. The options and prospects for updating these regulations are discussed.

9:35

4aNS4. Update on regulations adding noise to hybrid and electric cars. Dennis Weidemann (2633 Granite Rd., Fitchburg, WI 53771, dweid@mac.com) and Leslie D. Blomberg (Noise Pollution Clearinghouse, Montpelier, VT)

The United States National Highway Traffic Safety Administration in nearing and may have completed is rulemaking concerning adding noise to hybrid and electric cars by May 2014. This paper will examine what has happened in early 2014 with respect to these regulations. Updates to international regulations will also be presented.

9:55

4aNS5. Simulated and laboratory models of aircraft sound transmission in residences. Ashwin Thomas, Erica Ryherd, Thomas Bowling (Woodruff School of Mech. Eng., Georgia Inst. of Technol., c/o Erica Ryherd, Mech. Eng., Georgia Tech, Atlanta, GA, apthomas@gatech.edu), and Javier Irizarry (School of Bldg. Construction, Georgia Inst. of Technol., Atlanta, GA)

Current aircraft noise guidelines are based primarily on outdoor sound levels. As people spend the majority of their time indoors, however, human perception is highly related to indoor sound levels. Investigations are being made to provide further insight into how typical residential constructions affect indoor sound levels. A pilot, single-room "test house" has been built using typical mixed-humid climate region construction techniques and the outdoor-to-indoor transmission of sound was directly measured—with specific focus on continuous commercial aircraft signatures. The measurements included a variety of construction iterations (e.g., window type, wall construction) and a variety of instrumentation iterations (e.g., source and sensor locations). The results of this study are being used to validate and improve modelling software that simulates a wide range of construction types and configurations for other US climate regions. Overall, the project intends to improve the ability to predict acoustic performance for typical US construction types as well as for possible design alterations for sound insulation.

10:15–10:30 Break

Contributed Papers

10:30

4aNS6. Spatial regression relations between urban forms and road-traffic noise. Seo I. Chang (Environ. Eng., Univ. of Seoul, 163 Seoulsiripdaero, Dongdaemun-gu, Seoul 130-743, South Korea, schang@uos.ac.kr) and Bum Seok Chun (Ctr. for GIS, Georgia Inst. of Technol., Atlanta, GA)

Recent development of noise mapping tools allows us to generate sophisticated environmental noise maps where complicated acoustic phenomena including reflection by building facades, diffraction by horizontal and vertical edges of a building, and absorption by pavements can be considered with high level of accuracy. Therefore, if we have a noise map of an existing city and plan to do minor modifications, such as adding lanes to a road or locating new residential buildings along highways, we can assess and mitigate the induced impacts by simulating upon the existing noise map, e.g., installation of noise barriers or control of traffic flows. But, if a totally new city is built separately, what and how can we plan about the environmental noise? How can we do city-planning based on minimum information? What minimum information should be provided? Identification of the relations between urban forms and environmental noise can be helpful to city-planners at very early stage of planning. We performed spatial statistical analysis of road-traffic noise and urban forms by utilizing a GIS tool. Urban forms in the spatial regression model include residential and employee populations, building forms, traffic properties, and land-use pattern.

10:45

4aNS7. The spectral effect of masking of intruding noise by environmental background-noise. Giora -. Rosenhouse (Swantech, 89 Hagalil Str., Haifa 3268412, Israel, fwamtech@bezeqint.net)

Annoyance by noise depends strongly on its informative, spectral contents and individual effect on people. Yet, standards dictate certain formal limitations, ignoring such details. In practice, it happens in many cases of recreational areas, industrial premises and other kinds of activities, that even when results of measurements satisfy the standards limits, complaints do not stop, yielding threats of legal acts. Case studies of the effect, based on actual acoustic measurements are analyzed here, showing factors that cause extreme sensitivity to certain noise patterns, even if the total amount of noise remains unchanged. The effect of color difference is enhanced if the added noise has a certain periodicity, located where the background noise has lower masking effect. Since in many cases the background noise has less effect or resembles white or pink noise, certain noise sources can be clearly heard, if they include higher local amplitudes in the frequency spectra domain of the background noise. Acoustic solutions include means for undesired noise reduction to levels much below the background noise¸ by as much as by 9 dB, to allow background noise masking of disturbing sources. Such reduction alters its status from being strongly heard to the privacy zone.

11:00

4aNS8. Active control of traffic noise radiation and propagation. Qi Hu and Shiu-keung Tang (Dept. of Bldg. Services Eng., The Hong Kong Polytechnic Univ., Hung Hom, KLN, Hung Hom Na, Hong Kong, qi.bs.hu@connect.polyu.hk)

Active noise control tends to be challenging, especially for open space scenario. This work intends to control road traffic noise that is treated as an ideal line source with finite length, actively through the introduction of an array of secondary point monopole sources to modify the original sound field, which accordingly creates a quiet zone for the noise sensitive receivers. Three dimensional analytical formulation and numerical simulation are performed to compare the difference before and after the introduction of control sources, through which the optimal position and strength of each control source are studied.

11:15

4aNS9. Investigating human annoyance thresholds of tones in noise from a dose-response relationship. Joonhee Lee, Jennifer M. Francis, 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)

Noise with prominent tones from building mechanical systems is often detrimental to the environmental quality and leads to complaints. Previous studies have investigated the relationship between existing tonal noise metrics and human annoyance perception, but little is known about at what level the tones at assorted frequencies induce human annoyance. This paper investigates human annoyance responses due to noise with tones to produce a dose-response relationship for estimating the thresholds of annoyance to tones in noise. The subjective test is conducted using noise signals with varied loudness and tonalness thorough an Armstrong i-Ceiling system in the Nebraska indoor acoustic testing chamber. Binary logistic multiple regression models are used to predict the percentage of annoyed people or likelihood-to-complain with confidence intervals. This paper also examines the statistical performance of models with assorted noise metrics and non-acoustical variables to calculate the probability of occupants feeling annoyed for any given background noise with tonal components.

11:30

4aNS10. Differences between sound pressure levels measured outdoors in three heights commonly used in environmental noise impact assessment. Olmiro C. de Souza (UFSM, Undergraduate Program in Acoust. Eng., DECC-CT-UFSM, Av. Roraima 1000, Camobi, Santa Maria, RS 97105-900, Brazil, Acampamento, 569, Santa Maria, Santa Maria 97050003, Brazil, olmirocz.eac@gmail.com), Stephan Paul, and Diego Garlet (UFSM, Undergraduate Program in Acoust. Eng., DECC-CT-UFSM, Av. Roraima 1000, Camobi, Santa Maria, RS 97105-900, Brazil)

In environmental acoustics, outdoor sound pressure level can be measured at different heights over ground. ISO 1996 recommends measurements at 4 m, which is representative for multi-storey residential areas and in accordance with EU directives for noise map modeling. For other areas ISO 1996 recommends measurements 1.5 m over ground, a height that corresponds to the median height of adults ears. In Brazil, 1.2 m are commonly used for outdoor measurements and noise map calibrations as this height is recommended by the Brazilian standard NBR 10.151-2000. The goal of this work is to investigate the relationship between SPL measurements obtained at these three heights. Measurements were taken close to roads at a university campus, roads that in some cases have high traffic flows. The differences between measured A-weighted equivalent SPLs (LAeq) at the different heights were statistically analyzed. Difference distributions were found to be closely to normal distribution with some outliers. Mean values of the SPL differences remained below 3 dB. From the data obtained, it seems acceptable to calibrate a noise map model at a different height from the measured one using the mean difference as a correction term.

Session 4aPA

Physical Acoustics: Acoustic Radiation Forces, Streaming, and Applications

Bart Lipkens, Chair
Mech. Eng., Western New England Univ., 1215 Wilbraham Rd., Box S-5024, Springfield, MA 01119

Contributed Papers

8:30

4aPA1. Prediction of acoustic radiation forces in three dimensional flow through resonators. Ari Mercado and Bart Lipkens (Mech. Eng., Western New England Univ., 1215 Wilbraham Rd., Springfield, MA 01119, a.mercado@fdsonics.com)

In large scale acoustophoretic particle separation systems, the acoustic radiation force exerted on the particles exceeds the combined effect of the fluid drag force and gravitational force on the particle. This results in the trapping of the particles in the acoustic standing wave, followed by aggregation of the particles, and ultimately gravitational settling and separation of the secondary phase. The separation system typically consists of a flow chamber in which a three dimensional acoustic standing wave is generated by piezoelectric transducers. Accurate prediction models of the acoustic radiation force are needed so that they can be used as a tool in the design and development of such separation systems. The prediction model consists of two steps. First, COMSOL Multiphysics™ software is used to predict the acoustic field in the separation devices. Next, theoretical models [Gor'kov, Sov. Phys. Dokl. **6**, 773–775 (1962) and Ilinskii *et al*., J. Acoust. Soc. Am. **133**, 3237 (2013)] are used to calculate the acoustic radiation force on a suspended particle. Numerical results were verified by comparison with the theoretical results for a rectangular cavity [Barmatz and Collas, J. Acoust. Soc. Am. **77**, 928 (1985)]. [Work supported by NSF PFI:BIC 1237723.]

8:45

4aPA2. Design of a multi-element transducer for large volume acoustophoretic phase separation. Jason P. Dionne (FloDesign Sonics, Inc., 499 Bushy Hill Rd., Simsbury, Connecticut 06070, j.dionne@fdsonics.com) and Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA)

Efficient separation technologies for multi-component liquid streams that eliminate waste and reduce energy consumption are needed. In previous experiments around this novel platform technology, a single element transducer has been used to generate a high intensity three-dimensional ultrasonic standing wave resulting in an acoustic radiation force that is larger than the combined effects of fluid drag and buoyancy. Acoustic trapping of particles followed by enhanced gravitational settling is used to separate the secondary phase. A typical transducer is made of a PZT-8 2-MHz ceramic. This work reports on the comparison of the performance of a single element transducer to that of a multi-element transducer. Parametric simulation studies of multi-element transducer designs were performed to accurately predict the acoustic pressure field in the fluid flow with the goal of generating large acoustic radiation forces to assist in phase separation. COMSOL Multiphysics™ was used to run simulations and results were compared to an experimental prototype that consisted of a 2-in. by 1-in. flow chamber driven by a 1-in. by 1-in. 2-MHz transducer. The designs of the multi-element transducers consisted of two PZT-8 2-MHz transducers; one consisting of 16 elements and another of 25 elements. [Work supported by NSF PFI:BIC 1237723.]

9:00

4aPA3. Yeast filtration using large volume flow rate acoustophoretic separation. Brian McCarthy, Ben Ross-Johnsrud (FloDesign Sonics, 380 Main St., Wilbraham, MA 01095, b.mccarthy@fdsonics.com), and Bart Lipkens (Mech. Eng., Western New England Univ., Springfield, MA)

Cell processing occurs in many technologies such as lab-on-a-chip, biopharmaceutical manufacturing, and food and beverage industry. Centrifuges and filters are used in preprocessing and filtration stages. These technologies are not continuous flow filtration methods, a drawback for automation and miniaturization. Continuous cell filtration using ultrasonic standing waves has been successfully used at limited flow rates [Hawkes and Coakley, Enzyme Microbial Technol. **19**, 57–62 (1996)]. Advantages of ultrasonic particle filtration are continuous operation with no mechanical moving parts, no risk of membrane fouling, and no consumables. We present a novel design of an acoustophoretic particle separation system operating at large volume flow rates. The technology operates by creating ultrasonic standing waves that produce an acoustic radiation force on particles which exceeds the drag and gravitational forces thereby trapping the particles. Over time aggregation of trapped particles results in gravitational settling of the agglomerated particles. The system comprises a 1 in. $\times$ 1 in. flow section and is powered by a 2 MHz PZT-8 transducer and typically operates at flow rates up to 2 L/H. Concentration reductions in excess of 90% are obtained for yeast suspensions of rehydrated *S. cerevisiae* in RO-DI water with volume concentrations ranging from 0.5 to 3%. [Work supported by NSF PFI:-BIC 1237723.]

9:15

4aPA4. The role of bubbles in the atomization of liquids and tissues. Julianna C. Simon (Appl. Phys. Lab., Ctr. for Industrial and Med. Ultrasound, Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105, jcsimon@uw.edu), Oleg A. Sapozhnikov, Vera A. Khokhlova (Appl. Phys. Lab., Ctr. for Industrial and Med. Ultrasound, Univ. of Washington, Seattle, WA, and Dept. of Acoust., Phys. Faculty, Moscow State Univ., Seattle, WA and Moscow, Russian Federation), Yak-Nam Wang, Wayne Kreider, Lawrence A. Crum, and Michael R. Bailey (Appl. Phys. Lab., Ctr. for Industrial and Med. Ultrasound, Univ. of Washington, Seattle, WA)

Ultrasonic atomization, or the emission of droplets from a liquid exposed to air, has been studied for many decades. The most accepted theory of atomization, the cavitation-wave hypothesis, states that droplets are emitted by a combination of capillary wave instabilities and cavitation bubble collapses. Recently, it was shown that tissues could also be atomized and that the result of atomization was surface erosion. Using a high static pressure chamber, we investigated the role of bubbles in the atomization of tissues and liquids. A 2-MHz, aluminum-lensed transducer was focused at the surface of either water or *ex vivo* bovine liver. In water at 1200 W/cm^2 ($p_+ = 6.8$ MPa, $p_- = 5.3$ MPa), we found that atomization ceased at an overpressure of 6.9 MPa, yet droplets were again released when the static pressure was increased to 13.8 MPa. In tissue at a linear *in situ* intensity of

22 000 W/cm^2 ($p_+ = 67$ MPa, $p_- = 16$ MPa), we found that a small increase in the static pressure (1.4 MPa) produced a qualitative change in atomization and caused thermal denaturation of the fractionated tissue rather than ejection from the surface. [Work supported by NIH EB007643, NIH DK043881, and NSBRI through NASA NCC 9-58.]

9:30

4aPA5. Optical theorem for beams and application to radiation forces and torques by Bessel beams. 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)

The optical theorem is known as one of the central theorems in scattering theory; the theorem for plane waves relates the extinction by an object to the scattering amplitude at the forward direction. For a general non-diffracting beam an extended theorem was given recently [Zhang and Marston, J. Acoust. Soc. Am. **131**, EL329–EL335 (2012)]. The theorem relates the extinction to the scattering amplitude at the forward direction of plane wave components of the invariant beam. The theorem was used to examine the extinction by a sphere centered on the axis of a non-diffracting Bessel beam [Zhang and Marston, Bio. Opt. Express **4**(9), 1610–1617 (2013)]. The results are applied to recover axial radiation force [Zhang and Marston, Phys. Rev. E 84, 035601(R) (2011)] and torque [Zhang and Marston, Phys. Rev. E **84**, 065601(R) (2011)] exerted by the Bessel beam on the sphere. This form of optical theorem may be extended to a broader class of incident wave fields. [Zhang was supported by the 2013-14 ASA F. V. Hunt Postdoctoral Research Fellowship. Marston was supported by ONR.]

9:45–10:15 Break

10:15

4aPA6. A sonic levitation system for the study of Faraday waves on bubbles. Jorge Escobedo and R. Glynn Holt (Dept. of Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, jorge189@bu.edu)

Acoustic levitation is a method by which the gravitational force on a sample can be balanced by the time-averaged acoustic radiation force in a standing wave. Levitation has in the past been utilized as an ideal system to study phenomena at fluid interfaces, since boundary influences are small. Previous work on Faraday waves on bubbles has been carried out at ultrasonic frequencies, where the disadvantages are the need for fast diagnostics at small spatial scales. In this talk we describe efforts to develop a sonic frequency levitation system, employing hardware elements from previous NASA investigations in drop physics. Proof-of-concept styrofoam levitation in air will be demonstrated, and injection and deployment schemes for large (1-in. diameter) bubbles will be discussed. [Work supported by the Robert W. Young Undergraduate Award of the ASA.]

10:30

4aPA7. Experimental investigation of acoustic streaming flows inside a standing wave tube. Yasser Rafat, Shahin Amiri, Rani Taher, and Luc Mongeau (Mech. Eng., McGill Univ., 845 Sherbrooke St. West,, Montreal,, QC H3A 0G4, Canada, yasser.rafat@mail.mcgill.ca)

Acoustic streaming is identified as one of the several phenomena which affect the efficiency of thermoacoustic system by causing convective heat transfer. In the context of thermoacoustic machines, most of the experimental and numerical studies were performed on Rayleigh acoustic streaming. In the present study, different acoustic streaming flows within a standing wave tube were investigated. Experiments were performed using particle image velocimetry. A rectangular Plexiglas resonator was used as an idealized standing wave thermoacoustic refrigerator. The experimental results were compared with linear acoustic theory to ascertain their validity. Acoustic streaming generated due to interaction of standing wave with thermoacoustic core was also studied. Simplified components were used to model the stack and heat exchangers. It was found that presence of rigid obstacles in the standing wave resonator changed the streaming flow completely. Both the magnitude and shape of the streaming cells changed when compared with the classical Rayleigh streaming cell. The resulting local streaming velocity due to rigid obstacles in the standing wave tube had very high magnitude when compared with streaming in an empty standing wave tube.

10:45

4aPA8. Acoustic streaming from a resonant elastic surface vibration. Megha Sunny (Carnegie Mellon Univ., Lowell, Massachusetts), Katherine Aho, John C. Miniter, and Charles Thompson (Univ. of Massachusetts Lowell, 1 University Ave., Lowell, MA 01854, katherine_aho@student.uml.edu)

Acoustic streaming induced by the vibration of an elastic membrane is examined. Adjustment of the impedance along the membrane allows one to control the spatial characteristics of the time-averaged fluid motion. It is shown that under resonant conditions spatial localization of the time-averaged Reynolds stress occurs. As a result, the fluid motion outside of the viscous boundary layer is driven into motion by a wall-jet at locations of maximal vibration. The velocity field is evaluated in terms of elemental Stokeslets where streaming motion is expressed in terms of a surface force distribution. At high oscillatory Reynolds numbers, regularization procedures to deal with singularities that occur in the formulation are addressed. Streaming motion is expressed in terms of the surface force distribution. Results for oscillatory and time-averaged fluid motion are presented.

11:00

4aPA9. Interaction acoustic radiation force and torque on a cluster of spheres suspended in an inviscid fluid. Jose Henrique A. Andrade (Physical Acoust. Group, Inst. of Phys., Federal Univ. of Alagoas, conjunto pau darco rua d 23, maceio 57043394, Brazil, henriquealopes@gmail.com), Mahdi Azarpeyvand (Mech. Eng., Univ. of Bristol, Bristol, United Kingdom), and Glauber T. Silva (Physical Acoust. Group, Inst. of Phys., Federal Univ. of Alagoas, Maceió, AL, Brazil)

The acoustic radiation force and torque exerted by a time-harmonic beam of arbitrary wavefront on a cluster of suspended spheres in an inviscid fluid is theoretically analyzed. In the proposed method, the effective incident wave is modelled as a coherent sum of an external beam and the contributions from the re-scattering events by other spheres present in the medium. Using the translational addition theorem for spherical functions the effective beam-shape and scattering coefficients are numerically computed [J. Acoust. Soc. Am. **98**, 495 (1995)] for different external incident fields. The radiation force and torque exerted on the probe sphere can then be calculated using the farfield partial-wave expansion method [J. Acoust. Soc. Am. **130**, 3541 (2011); Europhys. Phys. Lett. **97**, 54003 (2012)]. The method was employed to obtain the radiation force due to an external plane and spherical waves on a cluster of three solid elastic or fluid spheres suspended in water. The results show that the radiation force deviates considerably from that exerted solely by the external incident wave and that the radiation torque arises on the spheres when a asymmetric spatial distribution of the effective incident acoustic field takes place in the medium. In addition, the proposed method may help on the study of acoustic tweezers devices and acoustofluidic systems, which involve several suspended particles.

11:15

4aPA10. Numerical study of Rayleigh-type acoustic streaming based on a three-dimensional incompressible flow model. Takeru Yano (Osaka Univ., 2-1, Yamada-oka, Suita 565-0871, Japan, yano@mech.eng.osaka-u.ac.jp)

Rayleigh-type acoustic streaming induced by a plane standing wave in a rectangular parallelepiped is numerically studied on the assumption that the streaming motion is an incompressible flow and induced by the so-called limiting velocity on the outer edge of the acoustic boundary layer on the wall of the rectangular parallelepiped. Solving the three-dimensional incompressible flow equations with a standard finite difference scheme, we show that the streamline indicates chaotic behaviors even when the streaming velocity field converges to a time-independent state (steady flow) for moderate Reynolds numbers. Based on the result, we can discuss an efficiency of mixing by the time-independent Rayleigh-type acoustic streaming motions in three-dimensional boxes.

Session 4aPP

Psychological and Physiological Acoustics and Speech Communication: Cambridge Contributions to Auditory Science: The Moore—Patterson Legacy

Andrew J. Oxenham, Cochair
Psychology, Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455

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

Robert P. Carlyon, Cochair
MRC Cognition & Brain Sciences Unit, 15 Chaucer Rd., Cambridge CB1 3DA, United Kingdom

Christopher Plack, Cochair
School of Psychological Sciences, Univ. of Manchester, Ellen Wilkinson Bldg., Oxford Rd., Manchester M13 9PL, United Kingdom

Chair's Introduction—8:00

Invited Papers

8:05

4aPP1. Psychophysics to the rescue! Translational hearing research in Cambridge. Robert P. Carlyon (MRC Cognition & Brain Sci. Unit, 15 Chaucer Rd., Cambridge CB1 3DA, United Kingdom, bob.carlyon@mrc-cbu.cam.ac.uk)

Roy Patterson and Brian Moore, perhaps more than any other psychoacousticians, have succeeded in applying their research for the common good. Those who have benefited from this translation of basic research include users of hearing aids and of auditory warnings. I will describe the results of recent experiments aimed at improving hearing by another group, namely users of cochlear and auditory brainstem implants. These include attempts to exploit the polarity sensitivity of the electrically stimulated auditory system in order to extend the ranges of pitch that can be conveyed by each type of implant.

8:25

4aPP2. The auditory image model and me. Michael Akeroyd (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)

As a Ph.D. student in Roy Patterson's group at the MRC Applied Psychology Unit in Cambridge, UK, in the early 1990s, I was introduced to computer models of hearing. Much of my Ph.D. was devoted to exploring Roy's Auditory Image Model, a time-domain model for representing regularities in hearing sensations that we hear. It was built on a gammatone filterbank, a hair-cell model, and strobed temporal integration, and was programmed with a speed that was remarkable for the age. The pictures and movies that it made—and the insights into hearing that it gave—were exciting and inspiring; my resulting enthusiasm for what good models can do has remained with me throughout my scientific career. This talk will describe some of Roy's contributions to modelling and his influence on the field, as ever-improving computational models are crucial to making progress in understanding how hearing works. [Work supported by the Medical Research Council and the Chief Scientist Office, Scotland.]

8:45

4aPP3. Acoustic surface structure, across-formant integration, and speech intelligibility. Brian Roberts, Robert J. Summers (Psych., School of Life & Health Sci., Aston Univ., Birmingham B4 7ET, United Kingdom, b.roberts@aston.ac.uk), and Peter J. Bailey (Dept. of Psych., Univ. of York, York, United Kingdom)

An important aspect of speech perception is the ability to group or select formants using cues in the acoustic surface structure—for example, fundamental frequency (F0) differences between formants promote their segregation. This study explored the role of more radical surface-structure differences. Three-formant (F1 + F2 + F3) synthetic speech analogues were derived from natural sentences. In one experiment, F1 + F3 were generated using second-order resonators (R1 + R3) and a monotonous glottal source (F0 = 140 Hz); in the other, F1 + F3 were tonal analogues (T1 + T3). F2 could take either form (R2 or T2). In some conditions, the target formants were

presented alone, either monaurally or dichotically (left ear = F1 + F3; right ear = F2). In others, they were accompanied by a competitor for F2 (F1 + F2C + F3; F2), which listeners must reject to optimize recognition. Competitors (R2C or T2C) were created using the time-reversed frequency and amplitude contours of F2. In the absence of F2C, the effect of surface-structure mismatch between F1 + F3 and F2 was typically modest. When F2C was present, intelligibility was lowest where F2 was tonal and F2C was a buzz-excited resonance, irrespective of which type matched F1 + F3. This finding suggests that surface structure type, rather than similarity, governs the phonetic contribution of a formant. [Work supported by ESRC.]

9:05

4aPP4. Enhancement of forward suppression begins in the ventral cochlear nucleus. Ian M. Winter (Cambridge Univ., The Physiological Lab., Downing St., Cambridge CB2 3EG, United Kingdom, imw1001@cam.ac.uk), Naoya Itatani (Univ. of Oldenburg, Oldenburg, Germany), Stefan Bleeck (Univ. of Southampton, Southampton, United Kingdom), and Neil Ingham (Kings College, London, United Kingdom)

A neuron's response to a sound can be suppressed by the presentation of a preceding sound (aka forward masking/forward suppression). Early studies in the auditory nerve have suggested that the amount of forward suppression was insufficient to account for behavioral data. Modeling studies have, however, suggested that forward suppression could be enhanced by coincidence detection mechanisms in the brainstem. Using a two-interval forced-choice threshold tracking algorithm, we compared forward suppression for different neuronal populations in the ventral cochlear nucleus (VCN) and the inferior colliculus of anesthetized guinea pigs. In both nuclei, onset-type neurons showed the greatest amounts of suppression (16.9–33.5 dB) and, in the VCN, these recovered with a faster time constant (14.1–19.9 ms). Neurons with sustained discharge demonstrated reduced suppression (8.9–12.1 dB) and recovery time constants of 27.2–55.6 ms. The growth of suppression, with increasing suppressor level, was compressive, but this compression was reduced in onset-type units. The threshold elevations recorded for most unit types were insufficient to account for the magnitude of forward masking as measured behaviorally; however, some units classified as onset responders demonstrated a wide dynamic range of masking, similar to that observed in human psychophysics.

9:25

4aPP5. Linear and log frequency rippled spectra. William Yost, Xuan Zhong, and Anbar Najam (ASU, PO Box 870102, Tempe, AZ 85287, william.yost@asu.edu)

Roy Patterson's Ph.D. dissertation investigated the pitch-shift of the residue and pitch perception has been a significant topic of interest to Roy ever since. I had the good fortune of collaborating with Roy on several studies involving pitch perception, especially on what we called regular interval stimuli (RIS), most notably iterated rippled noise (IRN). In addition to IRN stimuli which are characterized as stimuli with regularly spaced spectral peaks and valleys on a linear frequency axis, many studies have investigated stimuli that have regularly spaced spectral peaks and valleys on a logarithmic axis. Both set of stimuli produce a timbre/pitch-like sound quality. In some cases the sound quality of the two types of stimuli are difficult to perceptually separate. While RIS models of pitch processing (e.g., autocorrelation-based models) can account for many of the IRN pitch data, it is not clear what mechanisms produce the timbre/pitch-like qualities of log-frequency, rippled-spectra stimuli. The current paper involves three experiments designed to better understand auditory processing of rippled-spectra stimuli in order to determine if there may be some common perceptual elements that underlie the perception of such stimuli. [Research supported by the AFOSR.]

9:45–10:00 Break

10:00

4aPP6. Frequency selectivity and the auditory filter. Andrew J. Oxenham (Psych., Univ. of Minnesota, 75 E River Parkway, Minneapolis, MN 55455, oxenham@umn.edu)

Roy Patterson and Brian Moore laid the foundations for our modern understanding of frequency selectivity in the human auditory system and have defined and later refined the measurement and the modeling of the auditory filter. Although the auditory filter is a theoretical construct, the frequency selectivity it represents is thought to reflect the filtering in the cochlea. This talk will review recent work on comparing behavioral measures of the auditory filter with physiological measures of cochlear tuning in humans and other species. Although there are clear pitfalls in using linear systems analysis to characterize an inherently nonlinear system, such as the cochlea, the results suggest that the framework established by Patterson, Moore and their colleagues provides robust estimates of frequency selectivity that are consistent with more direct physiological measurements of cochlear tuning. [Work supported by NIH grant R01DC012262.]

10:20

4aPP7. The relationship between speaker size perception and the auditory filter. Toshio Irino (Faculty of Systems Eng., Wakayama Univ., 930 Sakaedani, Wakayama 640-8510, Japan, irino@sys.wakayama-u.ac.jp) and Roy D. Patterson (Ctr. for the Neural Basis of Hearing, Dept. of Physiol., Development and Neurosci., Univ. of Cambridge, Cambridge, United Kingdom)

When we hear a new voice on the radio, we can tell whether the speaker is an adult or a child. We can also extract the message of the communication without being confused by the size information. This shows that auditory signal processing is scale invariant, automatically segregating information about vocal tract shape from information about vocal tract length. Patterson and colleagues have performed a series of experiments to measure the characteristics of size/shape perception [e.g., Smith *et al.*, J. Acoust. Soc. Am. **117**(1), 305–318 (2005)], and provided a mathematical basis for auditory scale invariance in the form of the stabilized wavelet-Mellin transform (SWMT) [Irino and Patterson, Speech Commun. **36**(3–4), 181–203 (2002)]. The mathematics of the SWMT dictates the optimal form of the auditory filter, insofar as it must satisfy minimal uncertainty in a time-scale representation [Irino and Patterson, J. Acoust. Soc. Am. **101**(1), 412–419 (1997)]. The resulting gammachirp auditory filter is an asymmetric extension of the earlier gammatone auditory filter—one which can explain the level dependence of notched-noise masking. Thus, although it is not immediately intuitive, speaker size perception and auditory filter shape are both aspects of a larger, unified framework for auditory signal processing.

10:40

4aPP8. Novel paradigms to investigate temporal fine structure processing. Christian Lorenzi (Dept d'etudes cognitives, Ecole normale superieure, 29 rue d'Ulm, Paris 75005, France, lorenzi@ens.fr)

A wide range of evidence has been presented to support the idea that aging and cochlear hearing loss impair the neural processing of temporal fine structure (TFS) cues while sparing the processing of temporal-envelope (E) cues. However, the poorer-than-normal scores measured in tasks assessing directly TFS-processing capacities may partly result from reduced "processing efficiency." The accuracy of neural phase locking to TFS cues may be normal, but the central auditory system may be less efficient in extracting the TFS information. This raises the need to design psychophysical tasks assessing TFS-processing capacities while controlling for or limiting the potential contribution of reduced processing efficiency. Several paradigms will be reviewed. These paradigms attempt to either: (i) cancel out the effect of efficiency (leaving only the temporal factor), (ii) assess TFS-processing capacities indirectly via E-perception tasks where efficiency is assumed to be normal for elderly or hearing-impaired listeners, or (iii) assess TFS-processing capacities indirectly via E-perception tasks designed such that impaired listeners (i.e., elderly or hearing-impaired listeners) should outperform control listeners (i.e., young normal-hearing listeners) if aging or cochlear damage cause a genuine suprathreshold deficit in TFS encoding. Good candidates in this regard are interference tasks. Pilot data will be presented and discussed.

11:00

4aPP9. The temporal coding of pitch: Insights from human electrophysiology. Christopher Plack (School of Psychol. Sci., Univ. of Manchester, Ellen Wilkinson Bldg., Oxford Rd., Manchester M13 9PL, United Kingdom, chris.plack@manchester.ac.uk)

Brian Moore and Roy Patterson have made seminal contributions to our understanding of pitch perception, and in particular the use of temporal pitch information by the auditory brain. The pitch of sounds may be encoded, at least in part, by the tendency of neurons to phase lock to the temporal fine structure and/or envelope of basilar membrane vibration. Direct physiological measures of this mechanism are difficult in humans. However, the gross activity of neurons in the brainstem can be measured using electrophysiological techniques. The frequency-following response (FFR) is an electrophysiological measure of phase locking in the rostral brainstem. The FFR differs between musicians and non-musicians, and across linguistic groups, and is sensitive to short-term pitch discrimination training. These findings suggest that the FFR may reflect neural activity relevant to the encoding of pitch, although other results suggest that it may reflect basic peripheral encoding, rather than the output of a pitch extraction process. Recent results from our laboratory show that combining behavioral and FFR measures can provide insights into the coding of the pitch of pure tones and the coding of musical consonance. The FFR may be a blunt tool, but it provides information that cannot be obtained using other techniques, and this may be particularly useful in investigations of the effects of age and hearing loss on the neural coding of pitch.

11:20

4aPP10. Brain imaging the activity associated with pitch intervals in a melody. Roy D. Patterson (Physiol., Development and Neurosci., Univ. of Cambridge, Downing Site, Cambridge CB2 3EG, United Kingdom, rdp1@cam.ac.uk), Stefan Uppenkamp (Medizinische Physik, Universität Oldenburg, Oldenburg, Germany), Martin Andermann, and André Rupp (Sektion Biomagnetismus, Universität Heidelberg, Heidelberg, Germany)

Early attempts to locate brain regions involved in pitch processing employed sequences of notes with no pitch, fixed pitch, and melodic pitch. They revealed a region of Heschl's gyrus lateral to primary auditory cortex where sequences with pitch produced more activity than noise, and regions where melody produced more activation than fixed pitch (in planum polare and the superior temporal sulcus). Subsequent research has focused on the fixed pitch region in Heschl's gyrus and the degree to which the activity is pitch specific. Recently, MEG techniques have been developed to compare the responses to sequences of notes as they occur within bars of music, and to separate current sources associated with attention to melody. This paper illustrates how the techniques can be used to investigate the hierarchy of pitch and melody processing as it occurs in four bar phrases with brass instruments. The experiments show that a given note elicits a larger response when it is part of a melody and the increment is associated with a source beyond auditory cortex. The paper shows how we might track the responses to orchestral instrument sounds presented in a musical context as they proceed through auditory cortex and beyond the temporal lobes.

11:40

4aPP11. Loudness summation across ears for hearing-impaired listeners. Brian C. Moore and Brian R. Glasberg (Experimental Psych., Univ. of Cambridge, Downing St., Cambridge CB3 9LG, United Kingdom, bcjm@cam.ac.uk)

The summation of loudness across ears is often studied indirectly by measuring the level difference required for equal loudness (LDEL) of monaurally and diotically presented sounds. Typically, the LDEL is 5–6 dB, consistent with the idea that a diotic sound is about 1.5 times as loud as the same sound presented monaurally at the same level, as predicted by the loudness model of Moore and Glasberg [J. Acoust. Soc. Am. **121**, 1604–1612 (2007)]. One might expect that the LDEL would be smaller than 5–6 dB for hearing-impaired listeners, because loudness recruitment leads to a greater change of loudness for a given change in level. However, previous data from several laboratories showed similar LDEL values for normal- and hearing-impaired listeners. Here, the LDEL was measured for normal-hearing and hearing-impaired listeners using narrowband and broadband noises centered on a frequency where the latter had near-normal audiometric thresholds (500 Hz) and at a frequency where audiometric thresholds were elevated (3000 or 4000 Hz). The LDEL was similar for the two center frequencies for the normal-hearing listeners, but was smaller at the higher center frequency for the hearing-impaired listeners. The results were predicted reasonably well by the loudness model of Moore and Glasberg.

Session 4aSA

Structural Acoustics and Vibration and Physical Acoustics: Acoustics of Cylindrical Shells I

Sabih I. Hayek, Cochair
Eng. Sci., Penn State, 953 McCormick Ave., State College, PA 16801-6530

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

Chair's Introduction—8:55

Invited Papers

9:00

4aSA1. Simple models for linear and nonlinear modal vibration of circular cylindrical shells. Jerry H. Ginsberg (School of Mech. Eng., Georgia Inst. of Technol., 5661 Woodsong Dr., Dunwoody, GA 30338-2854, j.h.ginsberg@comcast.net)

Beginning shortly after the Second World War, ONR sponsored projects at Columbia University, New York, that analyzed vibration and shock response of shell structures. This work necessitated ingenuity because it preceded the advent of numerical modeling tools. A central theme was energy methods. This paper will review two works of this type. Baron and Bleich [J. Appl. Mech. **21**, 178–184 (1954)] used the mechanical energies of the Flugge-Byrne-Lur'ye shell theory and a fundamental property of the modal energies to obtain accurate formulas for the three vibration branches and associated modal displacements of a simply supported cylindrical shell. This author [Ginsberg, J. Appl. Mech. **40**, 471–477 (1973)] used these modes to analyze finite amplitude effects at resonances. Examination of the order of magnitude of terms in the mechanical energies led to identification of nonlinear mode coupling. The resulting differential equations for the modal coordinates were solved by a singular perturbation technique. The outcome was a set of algebraic equations for the nonlinear frequency response, and disclosure of the conditions under which an azimuthally symmetric response destablizes in favor of an antisymmetric response as a consequence of nonlinear coupling.

9:20

4aSA2. Scattering by a cylinderical shell buried in elastic sediment. Angie Sarkissian, Saikat Dey, Brian H. Houston (Code 7130, Naval Res. Lab., Code 7132, 4555 Overlook Ave. S.W., Washington, DC 20375, angie.sarkissian@nrl.navy.mil), and Joseph A. Bucaro (Excet, Inc., Springfield, VA)

Scattering results are presented for the case of cylindrical steel targets buried in elastic sediment with sound incident from the air above. The STARS3D finite element program recently extended to layered, elastic sediments is used to compute the scattering and the resulting normal displacement at the interface since the specific focus here is detection by systems which rely on monitoring the acoustic displacements or displacement-related entities at the fluid-sediment interface. Results are compared for the scattered field produced by the cylinder buried in layered elastic sediment versus in fluid sediment and for the scattered field of a buried cylindrical shell versus a buried solid cylinder. [This work was supported by ONR.]

9:40

4aSA3. Acoustic radiation from fluid-loaded cylindrical shells—A review. Joe M. Cuschieri (Lockheed Martin MST, 100 East 17th St., Riviera Beach, FL 33404, joe@cuschieri.us)

The acoustic radiation from fluid-loaded cylindrical shells received significant attention in the past. Reviewing the literature, the number of papers published in this area is significant. The body of work covers thin walled small diameter shells applicable to the sound transmission in pipes, to large diameter shells with internal stiffeners and bulkheads. Also, considered is the influence of full and partial compliant coatings. A significant portion of work was based on analytical techniques useful to understand the phenomena and some of the critical parameters. More recent, with the availability of more capable computers and modeling codes, the focus has been on application of these computational tools to solving field problems. This paper reviews some of the past computational work and how some applications evolved from this work. However when presented with actual submerged cylindrical structures with complex internals, while modeling is useful, when dealing with the acoustic radiation from shell like structures at medium to high frequencies modeling tools still cannot handle the full extent of the problem and the prevalent approach still relies on implementation of good engineering practice.

10:00

4aSA4. Quantitative ray methods for scattering by tilted cylindrical shells. Philip L. Marston (Phys. and Astronomy Dept., Washington State Univ., Pullman, WA 99164-2814, marston@wsu.edu) and Scot F. Morse (Div. of Comput. Sci., Western Oregon Univ., Monmouth, OR)

Starting with a review of ray methods and phenomena associated with high frequency scattering by spheres and cylindrical shells in water viewed broadside, generalizations to tilted shells will be summarized. These extensions were found to be useful for meridional as well as helical ray backscattering enhancements associated with leaky (or supersonic) waves on shells [Morse and Marston, J. Acoust. Soc. Am. **112**, 1318–1326 (2002); Blonigen and Marston, J. Acoust. Soc. Am. **112**, 528–536 (2002)]. For such enhancements Fermat's principle is useful for identifying ray paths of interest. In the case of helical waves (and in the broadside special case), the scattering amplitude can be expressed in terms of a Fresnel patch area where the guided wave is excited on the shell. Fresnel patches also give insight into the relatively large magnitude of meridional ray contributions. The coupling coefficient is proportional to the radiation damping of the leaky wave on the shell and in some cases it is necessary to take into account the anisotropy of the phase velocity. Computational benchmarks include scattering into the meridional plane by tilted infinite cylinders. Related phenomena include enhancements from subsonic guided waves and applications to sonar imaging and time-frequency analysis. [Work supported by ONR.]

10:20–10:30 Break

10:30

4aSA5. Response of a cylindrical shell with finite length ring stiffeners. Andrew J. Hull (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, andrew.hull@navy.mil)

This talk derives an analytical model of a thin shell of infinite extent that contains periodically spaced ring stiffeners that have finite length. The governing model of the system is the Donnell shell formulation affixed to ring stiffeners that are modeled as translational springs in the radial direction. The shell is excited by an external load that is harmonic in time and space. An orthogonalization procedure is developed and the resulting system equations are an infinite set of algebraic equations containing a diagonal matrix that represents the shell dynamics and a sparse matrix that contains permutations of the Fourier coefficients of the Heaviside step function that represent the stiffener forces. This matrix equation is truncated and inverted and yields a solution of the shell displacements. An example problem is formulated and the effects of the stiffeners on the system dispersion curves are illustrated.

10:50

4aSA6. Prediction of a body's structural impedance and scattering properties using correlation of random noise. Sandrine T. Rakotonarivo (Mechanics and Acoust., IUT GMP Aix-En-Provence, Université de Aix-Marseille, Marseille, France, sandrine.rakotonarivo@univ-amu.fr), W. A. Kuperman (Marine Physical Lab., Scripps Inst. of Oceanogr., La Jolla, CA), and Earl G. Williams (Acoust., Naval Res. Lab., Washington, DC, DC)

The structural or surface impedance matrix (or equivalently the inverse of the structural Green's function) for an elastic body can be obtained by placing it in an encompassing and spatially random noise field and cross-correlating pressure and normal velocity measurements taken on its surface. The derived theory shows that the correlation method produces the exact analytic form of the structural impedance matrix. A numerical experiment is presented determining the structural impedance matrix of an infinite cylindrical shell excited by a spatially random noise field. These results are then used to compute the scattered field from a nearby point source, which is in agreement with known results. [Work supported by the Office of Naval Research.]

11:10

4aSA7. Acoustic directional response of distributed fiber optic sensor cables. Jeffrey E. Boisvert (NAVSEA Div. Newport, 1176 Howell St., Newport, RI 02841, jeffrey.boisvert@navy.mil)

Distributed fiber sensor systems based on Rayleigh backscatter interferometry have demonstrated the capability for highly sensitive strain measurements over tens of kilometers using low cost fiber-optic cable for terrestrial and maritime applications. These cables are typically multilayered in construction and contain a fiber optic (glass) core. A generalized multi-layered infinite-length cable is modeled using the exact theory of three-dimensional elasticity in cylindrical coordinates. The cable is excited by an acoustic plane wave with an arbitrary angle of incidence. At each angle of incidence the radial and axial strains within the cable are integrated over a desired sensor zone length to determine the optical phase sensitivity using an equation that relates the strain distribution in an optical fiber to changes in the phase of an optical signal. Results for the cable in a free-field water environment are presented for two different cable geometries. The analytical model was used to identify the root cause of a marked increase in cable sensitivity that exists at certain angles of incidence for plane wave excitation. Specifically, the enhancement occurs when the trace wavelength of the incident wave matches the propagation wavelength of a natural frequency of the cable. [Work supported by NAVSEA Division Newport ILIR Program.]

Contributed Papers

11:30

4aSA8. Real-time hybrid substructuring of a physical mass-spring system coupled to a fluid-loaded analytical substructure. Rui Botelho and Richard E. Christenson (Civil and Environ. Eng., Univ. of Connecticut, 261 Glenbrook Rd. Unit 3037, Storrs, CT 06269, rui.botelho@uconn.edu)

Real-time hybrid substructuring (RTHS) is a relatively new method of vibration testing that allows a coupled dynamic system to be partitioned into separate physical and numerical components or substructures. The physical and numerical substructures are interfaced together in real-time as a closed-loop hybrid experiment similar to hardware-in-the-loop (HWIL) testing, whereby the physical substructure is tested concurrently with a numerical simulation of the remaining system. This work describes uniaxial RTHS testing at the University of Connecticut Structures Research Laboratory applied to simplified fluid-loaded structural systems. These tests use a physical one degree of freedom (DOF) mass-spring system coupled to a fluid-loaded analytical substructure. One test uses a fluid-loaded plate as the analytical substructure, while another test uses a fluid-loaded cylinder. An overview of RTHS is also presented, including the details of the feedback control architecture for coupling physical and analytical substructures together using servo-hydraulic actuation with a model-based minimum-phase inverse compensation (MPIC) of the actuator dynamics. In addition, a convolution integral (CI) method for solving the fluid-loaded analytical substructures in real-time is described. Experimental results demonstrate that RTHS can accurately capture the dynamic interaction of a fluid-loaded structural system and provide physical insight into the coupled response.

11:45

4aSA9. Experimental research and analysis of the acoustical radiation of piezoelectric cylindrical transducers with various height-to-diameter aspect ratios. Corey Bachand (BTech Acoust. LLC, 151 Martine St., ATMC, Fall River, MA 02723, corey.bachand@cox.net), David A. Brown (ECE/ATMC, Univ. of Massachusetts Dartmouth, Fall, MA), and Boris Aronov (BTech Acoust. LLC, Fall River, MA)

Estimating the radiation characteristics of cylindrical transducers having moderate height-to-diameter aspect ratios (H/D $\approx$ 0.2–2) over a wide frequency range is subject to considerable error with closed-form analytical solutions. It is often the case that transducers for acoustic communication fall within this range of aspect ratios. Thus, most often numerical techniques are required to solve the acoustical radiation problem, particularly for cylinders where the surface configuration (end caps and curved walls) does not allow for separation of variables in the Helmholtz equation describing the acoustic pressure. Results of calculating radiation characteristics of finite-height cylinders based on a numerical technique developed by Kozyrev and Shenderov [Sov. Phys. Acoust. **23**(6), 230–236 (1980)] are presented. Several prototype piezoelectric cylindrical transducers with aspect ratios ranging from 0.3 to 1.0 were constructed as part of the research on the radiation characteristics of finite-height cylinders. The two cases of an air-backed internal cavity with shielded end caps and of a fluid-filled internal cavity without end caps are considered. Analytical and numerical radiation estimations are compared to measured results with the prototypes, and applicability of the analytical models for different aspect ratios and wave dimensions are discussed.

THURSDAY MORNING, 8 MAY 2014 EAST PREFUNCTION, 8:00 A.M. TO 12:00 NOON

Session 4aSC

Speech Communication: Cross-Language Topics in Speech Communication (Poster Session)

Megan Reilly, Chair
Dept. of Cognitive, Linguist., and Psychological Sci., Brown Univ., 190 Thayer St., Providence, RI 02912

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10.00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

4aSC1. Tone as a primary perceptual cue in cross-linguistic speech perception: A comparison of Cantonese and Mandarin second-language speech of English clusters. Yizhou Lan (Dept. of Chinese, Translation and Linguist, City Univ. of Hong Kong, Tat Chee Ave., Kowloon, Hong Kong, eejoe.lan@gmail.com)

The study examined the patterns of production and perception of L1 Cantonese and Mandarin speakers for English consonant-/l/ clusters in five consonant conditions (/ph, th, kh, f, s/) and three vowel conditions (/i, a, u/). Five Cantonese, 5 Mandarin, and 5 native English speakers were assigned to read aloud words in C[l]V and C[l]VC structure in carrier sentences. Results showed that Cantonese speakers' /l/ was often reduced, indicated by a shorter average duration of the CV transition compared with native English speakers. However, Mandarin speakers showed a longer duration in the same measurement. Despite the identical segmental and syllable structures of these two languages in the involved words, realizations were different. Nevertheless, we found pitch patterns of Mandarin speech of L2 English featured falling tone, whereas Cantonese speakers utilized level tones. To further examine the tonal effect, we normalized the tone from the production results to a level tone at 200 Hz and presented them, together with productions with real inserted vowels in between C and /l/, to another group of Mandarin and Cantonese speakers to discriminate in an ABX paradigm. Mandarin speakers scored a significantly lower accuracy rate, indicating that their perception of duration was influenced by tone structure.

4aSC2. Contrastive apical post-alveolar and laminal alveolar click types in Ekoka !Xung. Amanda L. Miller (Linguist, The Ohio State Univ., 222 Oxley Hall, 1712 Neil Ave., Columbus, OH 43210-1298, amiller@ling.osu.edu) and Jeffrey J. Holliday (Second Lang. Studies, Indiana Univ., Bloomington, IN)

Ekoka !Xung has four contrastive click types—dental, alveolar, lateral, and "retroflex." We provide acoustic and ultrasound results of five speakers' productions of the typical alveolar click and the contrastive "retroflex" click. Ultrasound results show that the "alveolar" click is apical post-alveolar and the "retroflex" click is laminal alveolar. The burst duration of the post-alveolar click averages 12 ms which is "abrupt," while the burst duration of the

alveolar click averages 30 ms, which is "noisy." Mixed effects logistic regression models tested the effects of rise time and burst duration. Burst duration differed significantly among the two clicks ($p < 0.001$), while the effect of rise time was not significant. The ratio of energy in the click noise-bursts below 20 ERB to the energy above 20 ERB is between 1.0 and 1.5 for the post-alveolar click, but between 0.5 and 1.0 for the alveolar click. The ratio was a significant predictor of click type ($p = 0.014$). The highest concentration of energy for the post-alveolar click is between 12 and 18 ERB, while the highest concentration of energy in the alveolar click is between 25 and 30 ERB. We attribute the frequency difference to a larger lingual cavity volume in the post-alveolar click, and a smaller volume in the alveolar click.

4aSC3. Cross language speech-in-noise perception by early Spanish-English bilinguals and English monolinguals. Page E. Piccinini and Marc Garellek (Linguist, Univ. of California, San Diego, 9500 Gilman Dr., La Jolla, CA 92093-0108, ppiccinini@ucsd.edu)

Bilinguals have shown a hyper-awareness of fine phonetic detail in speech, while also sometimes losing out on higher-level syntactic and semantic information in speech-in-noise studies. This study seeks to determine how bilinguals process speech in noisy environments across different language contexts. Specifically, this study tests whether bilinguals utilize certain phonetic cues to access higher-level information. Two experiments will be conducted. First, to determine how bilinguals process speech in noise in different language contexts, early Spanish-English bilinguals and English monolinguals learning Spanish listened to sentences mixed with white noise in English, Spanish, and code-switching (English to Spanish and Spanish to English) contexts. Preliminary results suggest early Spanish-English bilinguals perform significantly above chance on word identification in all contexts, performing best in the Spanish context. The second experiment will determine specifically which noise types (lower versus higher frequency) are most detrimental to word identification. This in turn will suggest what kind of phonetic information is utilized most by bilinguals versus monolinguals. These results will aid our understanding of how bilinguals could use their hyper-awareness of phonetic detail to overcome difficulties in other aspects of processing.

4aSC4. Native language interference on the overnight consolidation of a learned nonnative contrast. Sayako Earle and Emily B. Myers (Speech, Lang., and Hearing Sci., Univ. of Connecticut, 123 Davis Rd., Storrs, CT 06268, frances.earle@uconn.edu)

In a prior investigation, discrimination of a trained nonnative (dental/retroflex, Hindi) contrast was mediated by different effects of overnight consolidation depending on the time of day of training. For individuals trained in the evening, sleep appeared to promote continued improvement in discrimination for ~24 h without further training. For participants who were trained in the morning, performance returned to baseline following the overnight session interval. A possible explanation for the lack of improvement in the morning training group is that incidental exposure to the alveolar /d/, the category in which dental/retroflex are considered allophones in English, throughout the daytime interval interfered with overnight consolidation of the nonnative variants. We tested this interpretation directly, by training all participants ($n = 44$) in the evening and assigning them to one of two conditions of interference: passive exposure to a stream of either 1500 /bV/ or /dV/ tokens immediately after training. We observed continuous improvement in discrimination for ~24 h in those who were exposed to /bV/ tokens, while those who were exposed to the /dV/ tokens did not improve. Our results support the interpretation that incidental exposure to English prior to overnight memory consolidation interferes with sleep-mediated improvement in discrimination of an L2 contrast.

4aSC5. Vowel systems of quantity languages compared: Arabic dialects and other languages. Judith K. Rosenhouse (Linguist, SWANTECH Ltd., 89 Hagalil St., Haifa 3268412, Israel, swantech@013.net.il), Noam Amir, and Ofer Amir (Commun. Disord., Tel-Aviv Univ., Tel-Aviv, Israel)

The acoustic phonetic features of colloquial Arabic vowel systems are still not entirely researched. This paper studies phonetic structure of several Arabic dialects and other languages. A basic issue is the fact that Arabic is a quantity language; but from the published literature we see that vowel systems of Arabic dialects differ in many acoustic details. We researched two colloquial Arabic dialects which are spoken in Israel, and hitherto not acoustically studied. These dialects constitute the axis around which we conducted the literature-based comparison with vowel systems of a few other Arabic dialects and other languages which share similar quantity features (i.e., long and short vowels). The study reveals similarities and differences in pitch (F0), the first three formants and duration. These differences appear between the two Arabic dialects spoken in Israel, between them and other Arabic dialects, as well as between non-Arabic languages (English, German, Swedish, and Hungarian). The findings of our study are discussed in relation with the questions of (1) vowel spaces of short and long vowels and (2) speaker's sex-dependent differences.

4aSC6. The articulation of lexical palatalization in Scottish Gaelic. Jae-Hyun Sung (Linguist, Univ. of Arizona , P.O. Box 210025, Tucson, AZ 85721, jhsung@email.arizona.edu), Diana Archangeli (Linguist, Univ. of Hong Kong, Hong Kong, Hong Kong), Ian Clayton (English, Boise State Univ., Boise, ID), Daniel Brenner, Samuel Johnston, Michael Hammond, and Andrew Carnie (Linguist, Univ. of Arizona , Tucson, AZ)

Scottish Gaelic (Gàidhlig, henceforth SG) exhibits a rich system of consonant mutation, which is mostly governed by its morphology (Ladefoged *et al.* 1998; Gillies 2002; Stewart 2004). For instance, *bàta* "boat" changes to [v] when the word undergoes morphological inflection—e.g., *a bhàta* "his boat", in which the sound spelled *bh* is pronounced as [v]. Using ultrasound imaging, the present study investigates palatalization in SG, which is considered as one of lexicalized consonant mutation types. Experimental data was collected in Sabhal Mòr Ostaig, a college on the Isle of Skye. Preliminary results show a clear sign of palatalization across different consonant types in palatalization environments (i.e., when morphologically conditioned), represented by higher tongue contours in the front region of tongue. While the articulatory distinction between plain and palatalized consonants is significant, different syllabic positions (i.e., word-initial vs. -final palatalization) often yield individualized patterns.

4aSC7. An acoustic-phonetic account of phonotactic perceptual assimilation. Eleanor Chodroff, Anthony Arnette, Samhita Ilango, 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 identified a coronal-to-dorsal 'perceptual assimilation' in which English and French listeners identify Hebrew word-initial /tl/ and /dl/ as beginning with /k/ and /g/, respectively (Hallé and Best, 2007). However, the acoustic-phonetic factors that contribute to this misperception have not been thoroughly identified, and previous results indicate that /tl/ is misperceived more often than /dl/—an asymmetry that is surprising on phonological grounds. The present study further explored this perceptual assimilation in two experiments with English listeners and Hebrew stop-liquid-vowel syllables ([t,k,d,g] × [l,ʁ] × [a,o,u]). The first experiment, which used the same stimuli as Hallé & Best, replicated previous findings, including the asymmetry between /tl/ and /dl/. The second experiment employed stimuli produced by a different native Hebrew speaker. While coronal-to-dorsal assimilation was observed, the previous /tl/-/dl/ asymmetry was not found: /dl/ was perceived as dorsal-initial somewhat more often than /tl/, suggesting that there can be no consistent phonemic or phonotactic explanation of the rate of assimilation. In support of a phonetic account, we find that misperception rates in both experiments are highly correlated ($r > 0.65$) with the stimulus-specific degree of anticipatory coarticulation of the lateral, as reflected in the spectral shape of the stop burst.

4aSC8. The effect of talkers' language dominance on subjects' speech production of sibilant fricatives. Ya-ting Shih (Teaching Chinese as a Second Lang., Chung Yuan Christian Univ. , 200 Chung Pei Rd., Chung Li 32023, Taiwan, ninashih1982@gmail.com)

This study investigates the effect of talkers' language dominance on subjects' sibilant production in a bilingual community. Guoyu (Taiwanese Mandarin) has 3 sibilants: alveolar /s/, retroflex /ʂ/ and alveolo-palatal /ɕ/, while Taiwanese (a Southern Min dialect) only has /s/, which is palatalized

before /i/. Previous studies have shown that Taiwanese-dominant speakers in Taiwan has a merged category of /s/, /ʂ/ and /ɕ/. In addition, they treat [s] and [ɕ] as allophones of /s/. On the other hand, Guoyu-dominant speakers have a more distinctive three-way contrast of sibilants. This study explores whether listening to talkers with different language dominances affects subjects' speech production of Guoyu sibilants. Two female talkers, one is Taiwanese-dominant and the other is Guoyu-dominant, recorded Guoyu words containing target sibilants in word-initial position with comparable vowels. Forty bilingual adults' productions were elicited in a repetition task blocked by talker. The spectral centroid is obtained from the middle 40ms of each sibilant, along with the onset F2 of the following vowel. The two acoustic measures were plotted against each other and separated by talker. Preliminary results show that these subjects' productions differ when prompted by different talkers. Additional statistical tests will be performed to explore this production difference.

4aSC9. Speech intelligibility across native and non-native accents: Accent similarity and electrophysiological measures of word recognition. Louise Stringer and Paul Iverson (Speech, Hearing and Phonetic Sci., UCL, Chandler House, 2 Wakefield St., London WC1N 1PF, United Kingdom, l.stringer.11@ucl.ac.uk)

The intelligibility of accented speech in noise greatly depends on the pairing of speaker and listener, where two important factors are a listener's familiarity with a speaker's accent and the acoustic similarity between their accents. In this study, we present patterns of the intelligibility of standard British English, Glaswegian English and Spanish English accents for British and high and low proficiency Spanish listeners. We predict intelligibility will correlate with acoustic-phonetic similarity across accent pairings, in line with previous findings. As such, findings are expected to provide further support that accent similarity can predict patterns of accent intelligibility, even if listeners have little experience of a speaker's accent. Electrophysiological measures relating to phonological and semantic integration stages of word recognition will also allow the investigation of the influence of accent on the time course of word recognition, which has not previously been directly compared to accent intelligibility or applied in studies of accent processing by non-native listeners. Findings will be discussed in the context of previous exploratory findings in this field, and in reference to other factors influencing accent intelligibility.

4aSC10. Non-native vowel processing as reflected by neuronal architecture: Dynamic causal modeling of the magnetic mismatch response. Georgina Oliver-Roth, Paul Iverson (Speech, Hearing and Phonetic Sci., Univ. College London, Wakefield St. 2, London WC1N 1PF, United Kingdom, g.oliver@ucl.ac.uk), Sundeep Teki, and Alexander Leff (Inst. of Cognit. Neurosci., Univ. College London, London, United Kingdom)

The aim of this study was to examine how auditory vowel processing by native (L1) and non-native (L2) speakers is reflected in their neuronal source architecture and in coupling between brain regions. We used the magnetic Mismatch Response/MMNm to test automatic brain responses to within- and between-category vowels with English controls and English L1/French L2 speakers with a varying range of L2 proficiency. Additionally, participants performed a range of behavioral tasks which targeted vowel perception (category discrimination and vowel identification) and production. The MEG data from this study was analyzed conventionally and with Dynamic Causal Modeling/DCM in order to determine neuronal sources and the dynamic source architecture in the L1 and L2 brain. In summary, the right hemisphere supported the left during L2 vowel processing in low ability L2 speakers. Performance in L2 vowel category discrimination was linked to the MMNm for a vowel distinction which was particularly difficult for the L2 speakers. The MMNm indicated whether a speech sound had gained phoneme status in an L2. DCM showed that there was no difference both architecturally and functionally between an L1 speaker's and a highly proficient L2 speaker's brain with regards to vowel processing.

4aSC11. A cross-linguistic study of lexical stress shifts in level 1 [+cyclic] derivations. Paul R. Keyworth (English, Saint Cloud State Univ., 3710 W. Saint Germain St., Apt. #234, Saint Cloud, MN 56301-7319, kepa1104@stcloudstate.edu)

Laboratory phonology has been widely employed to understand the interactional relationship between the acoustic cues of English Lexical Stress (ELS)—duration, fundamental frequency, and intensity. However, research on ELS production in polysyllabic words is limited, and cross-linguistic research in this domain even more so. Hence, the impacts of second language (L2) experience and first language (L1) background on ELS acquisition have not been fully explored. This study of 100 adult Mandarin (Chinese), Arabic (Saudi Arabian), and English (Midwest American) speakers examines their ELS productions in tokens containing seven different stress-moving suffixes; i.e., Level 1 [+cyclic] derivations according to lexical phonology. Speech samples were systematically analyzed using Praat and compared using statistical sampling. Native-speaker productions provided norm values for cross-reference to yield insights into the proposed Salience Hierarchy of the Acoustic Correlates of Stress (SHACS). The author recently reported the main findings which support the idea that SHACS exists in L1 sound schemes, and that native-like command of these systems can be acquired by L2 learners through increased L2 input. Other results are expected to reveal the role of tonic accent shift, the idiosyncrasies of individual suffixes, conflicts with standard dictionary pronunciations, and the effects of frequency perception scales on SHACS.

4aSC12. Effects of observing or producing hand gestures on non-native speakers' auditory learning of Japanese short and long vowels. Yukari Hirata (East Asian Lang. and Literatures, Colgate Univ., 13 Oak Dr., Hamilton, NY 13346, yhirata@colgate.edu), Spencer D. Kelly (Psych., Colgate Univ., Hamilton, NY), Jessica Huang, and Michael Manansala (East Asian Lang. and Literatures, Colgate Univ., Hamilton, NY)

This study examined whether auditory training coupled with hand gesture can improve non-native speakers' auditory learning of phonemic vowel length contrasts in Japanese. Hirata and Kelly (2010) found that observing hand gesture that moved along with the rhythm of spoken short and long vowels in Japanese did not uniquely contribute to non-native speakers' auditory learning. The present study compared effects of four types of training to examine whether there is a more effective method: (1) producing syllabic-rhythm gesture, (2) observing syllabic-rhythm gesture, (3) producing moraic-rhythm gesture, and (4) observing moraic-rhythm gesture. Each of native English speakers ($N=88$) participated in one of the four types of training in four sessions, and took a pretest and a posttest that measured their ability to auditorily identify the vowel length of novel words without hand gesture. Tested disyllable pairs had the contrast in the first and the second syllables, spoken in sentences at slow and fast rates. Results showed that all four groups improved significantly (9%), but the amount of improvement did not differ. However, 'observing syllabic-rhythm gesture' was the only condition in which auditory learning was balanced between the first and the second syllable contexts and between the slow and fast rates.

4aSC13. Perceptual learning of lexical tones by native speakers of English. Guannan Shen, Erika Levy, and Karen Froud (Teachers College, Columbia Univ., 509 West 122nd St., Apt. 18, New York, NY 10027, mandy.g.shen@gmail.com)

Whether native speakers of non-tonal languages can acquire categorical representations of lexical tones remains controversial. This study investigates the acquisition of lexical tone categories by native English speakers learning Mandarin Chinese as a foreign language by comparing the categorical perception of lexical tones between three groups of listeners: (1) native English speakers who had taken advanced Mandarin courses in colleges; (2) inexperienced native English speakers; and (3) native Mandarin speakers. Two tone continua derived from natural speech within carrier phrases were

created through interpolation within two tone contrasts (T1/T4; T2/T3). Assessments of categorical perception, including an identification task and a discrimination task, were conducted on all three groups of participants. Results showed classic categorical perception of tones by native Mandarin speakers. The inexperienced English speakers performed near chance on discrimination tasks and showed significantly broader identification boundaries. The learners of Mandarin showed similar categorical perception to native Mandarin speakers with comparable identification boundaries and discrimination scores. The results indicate that native speakers of non-tonal languages can learn to perceive lexical tones categorically. Experience-based perceptual categorization and acoustic cues for tonal language learners are discussed.

4aSC14. The effect of language experience on the ability of non-native listeners to identify Japanese phonemic length contrasts. Miwako Hisagi (Speech Commun. Studies, Iona College, 715 North Ave., New Rochelle, NY 10801, mhisagi@hotmail.com), Keiichi Tajima (Psych., Hosei Univ., Tokyo, Japan), and Hiroaki Kato (Universal Commun. Res. Inst., National Inst. of Information and Communications Technol. (NICT), Kyoto, Japan)

This study investigated how language experience affects second-language (L2) listeners' ability to perceive phonemic length contrasts in the face of stimulus variability. Native English-speaking learners of Japanese ($N = 42$) participated in an identification task in which the stimuli were Japanese words contrasting in vowel or consonant length, produced in isolation or embedded in a carrier sentence at slow, normal, or fast speaking rates, presented in a random order. Participants also received an Oral Proficiency Interview (OPI), developed by the American Council on the Teaching of Foreign Languages (ACTFL), to assess their Japanese proficiency on a 10-level scale. Results showed that identification accuracy as measured by d' was weakly correlated with OPI level ($r = 0.298$), and moderately correlated with number of semesters enrolled in Japanese language courses ($r = 0.401$). Speaking rate significantly affected performance, but its effect differed by context. For word-in-isolation context, d' was highest at the normal rate, while for word-in-sentence context, d' was highest at the slow rate. However, the effect of speaking rate was not reduced as a function of OPI level or number of semesters in Japanese courses. Thus, language experience may not always strongly predict L2 speech perception. [Work supported by JSPS-KAKENHI, MIT-RLE and -Linguistics & Philosophy.]

4aSC15. Can adjustment to accented speech affect native language perception? Eva M. Lewandowski (Psych., Emory Univ., 36 Eagle Row, Dept. of Psych., Ste. 270, PAIS Bldg., Atlanta, GA 30306, elewand@emory.edu), Teljer Liburd (Psych. and Learning Res. and Development Ctr., Univ. of Pittsburgh, Pittsburgh, PA), and Lynne C. Nygaard (Psych., Emory Univ., Atlanta, GA)

The human auditory system can quickly accommodate foreign-accented speech. However, the cognitive mechanisms underlying perceptual adjustment to non-native speech are not fully understood. The current study examined the perceptual consequences of adaptation toward foreign-accented speech on native language perception. Native English speakers performed an auditory shadowing task on word-length utterances in English. There were four blocks of trials. The words in the critical block (Block 3) were spoken by either a native American English speaker or a native Spanish speaker. The speaker in flanking blocks (Blocks 1–2, 4) was the same speaker, a different speaker with the same accent, or a different speaker with a different accent. Shadowing response times in the critical block were used to assess rapid perceptual adjustment and readjustment. Results showed that the nature of the preceding context influenced response times. Response times for items in the first quartile of the critical block were reliably slower when accent and talker changed than when accent and talker remained constant. These findings suggest that listeners develop perceptual expectations about ongoing speech, which when violated incur a short-term processing cost even for spoken words in the listeners' native language.

4aSC16. Word recognition in early bilingual adults for two degradation types. Rachel Shepherd and Tessa Bent (Dept. of Speech and Hearing Sci., Indiana Univ., 200 S. Jordan Ave., Bloomington, IN 47405, rachshep@imail.iu.edu)

Early bilingual adults have difficulty perceiving speech in noise compared to monolingual adults; however, the cause of the deficit is unknown. Further, the extent to which this deficit extends to other types of degradation, such as source degradation (e.g., a nonnative accent) has not been investigated. The current study investigated word recognition under environmental and source degradation by 24 monolingual and 24 bilingual listeners, who learned English and at least one other language before age 6. Participants identified sentences produced by one native and one nonnative talker in both quiet and noise-added conditions. Although noise was more detrimental to bilinguals than monolinguals, the presence of a nonnative accent caused a similar decline for both groups. Results from standardized tests of vocabulary, reading, spelling, nonverbal intelligence, and phonological processing showed two differences between the groups: bilinguals outperformed monolinguals on the nonverbal intelligence test (Raven's Standard Progressive Matrices) and bilinguals performed less accurately than monolinguals on the vocabulary assessment (Peabody Picture Vocabulary Test). Therefore, the speech-in-noise deficit for bilinguals may be traced to their weaker vocabulary knowledge. This study demonstrates that early bilinguals experience a word-recognition disadvantage under environmental degradation but not source degradation.

4aSC17. The production of non-modal phonation types in English vowels by Brazilian speakers. Ana Paula Engelbert (Head and Neck Surgery, UCLA, 1000 Veteran Ave., Los Angeles, CA 90095, anaengelbert@ucla.edu)

Esling (2000) claims that each language has its own pattern of physiological behavior in which articulators are trained to operate in different ways based on the language's phonetic structure. To test this claim, this study compares phonation types in speech production when Brazilians speak Portuguese and English. More specifically, we investigate coarticulation effects of consonants on vowels in English with regards of non-modal phonation. According to Garellek (2012), non-contrastive non-modal phonation happens in English vowels due to adjacent glottalized and aspirated consonants. However, this coarticulation effect does not happen in Brazilian Portuguese because voiceless stops have short lag VOT and neither voiced nor voiceless stops are allowed as codas. Thus, our hypothesis is that bilingual Brazilians do not produce non-modal phonation due to coarticulation when producing English vowels. To test this hypothesis, native speakers of English and Brazilians who are proficient speakers of English were recorded performing reading tasks in a soundproofed room. The target words containing the vowels to be measured were placed in a carrier-sentence. The acoustic analysis was based on H1*-H2*, H1*-A2*, and HNR measures. [Research funding by CAPES (Brazil) and Fulbright.]

4aSC18. Acquisition of the complex three-way Korean plosive contrast by native English speakers. Tyler Perrachione (Dept. of Speech, Lang., and Hearing Sci., Boston Univ., 635 Commonwealth Ave., Boston, MA 02215, tkp@bu.edu), Amy S. Finn, Jennifer Minas (McGovern Inst. for Brain Res., Massachusetts Inst. of Technol., Cambridge, MA), Caitlin Tan (Dept. of Brain and Cognit. Sci., Massachusetts Inst. of Technol., Cambridge, MA), Brian Chan, and John D. Gabrieli (McGovern Inst. for Brain Res., Massachusetts Inst. of Technol., Cambridge, MA)

Learning to perceive foreign language speech sounds is a core challenge in adult second language acquisition. Previous research has considered how listeners learn novel foreign language categories for a known phonetic continuum [e.g., voice onset time (VOT)], or how listeners learn to use a previously unattended phonetic feature (e.g., F3). We investigated perceptual learning of the Korean three-way plosive contrast (lenis, aspirated, and fortis) by native English speakers. Unlike VOT continua in other languages,

this contrast is distinguished by complex trading relations between VOT and pitch, with place of articulation differences in VOT adding further complexity. In this study, participants (N = 38) learned a vocabulary of 18 Korean pseudowords comprised of six minimal triplets (e.g., pan, ban, and ppan) by undergoing four days of high-variability (multi-talker) training on a lexical identification task. Mixture model analysis suggested two learner groups: (1) two-thirds of the participants were partially successful at learning words beginning with the fortis stops, but did not differentiate the lenis and aspirated stops; and (2) one-third of the participants successfully learned words beginning with the fortis stops, and exhibited progress distinguishing the lenis and aspirated stops. (Fortis stops most closely resembled listeners' existing English voiced stop categories.) Both groups acquired these contrasts best for bilabial stops and least accurately for alveolar stops.

4aSC19. Prosodic realization of focus in American English by Beijing Mandarin learners. Ying Chen (Dept. of Linguist, Univ. of Oregon, 124 Agate Hall, 1290 University of Oregon, Eugene, OR 97402, ychen12@uoregon.edu)

In addition to an increase of duration, F0 and intensity in phonetically realizing focus, post-focus compression (PFC) of F0 and intensity has been found in many languages, including American English and Beijing Mandarin. Recent studies found that PFC did not easily transfer from language to language (Wu and Chung, 2011); however, language experience impacted the realization of PFC (Chen *et al.*, 2012). The effect of length of residence (LOR) in an L2-speaking environment on L2 pronunciation accuracy remains controversial (Piske, 2007). The current study examined English focus production of Beijing Mandarin learners, who were college freshmen, residing in the United States for 3 to 6 months, and college seniors for 3.5 to 4 years. Compared to the control group, both learner groups produced comparable patterns of duration change; the freshman group did not present significant PFC of F0 and intensity in either initial-focus or medial-focus condition; the senior group presented a native-like PFC in the initial-focus condition and an intermediate pattern of PFC among the three groups in the medial-focus condition. The preliminary results indicate that Beijing Mandarin learners with long LOR in the US produced more native-like prosodic focus in English than those with short LOR in the United States.

4aSC20. Transfer effects in perception of a familiar and unfamiliar language. Charles B. Chang (Dept. of Linguist, Rice Univ., P.O. 1892, MS 23, Houston, TX 77251, cbchang@post.harvard.edu)

Second-language (L2) speech perception is typically worse than first-language (L1) perception, a disparity often attributed to negative transfer (interference) from the L1 of L2 listeners. The current study investigated the hypothesis that L1 transfer is not always negative, but variable depending on the nature of L1 perceptual biases. In Experiment 1, four groups of L2 English speakers whose L1s (Japanese, Korean, Mandarin, and Russian) differ in the relative informativeness of vowel-to-consonant transition cues were tested on their perception of English segments that rely crucially on these cues: final unreleased voiceless stops. In comparison to L1 English listeners, L1 Japanese, Russian, and Mandarin listeners performed significantly worse, whereas L1 Korean listeners performed significantly better. In Experiment 2, when the same groups were tested on similar Korean stimuli, L1 Russian listeners outperformed all other groups except the Korean group. These results provide evidence that L1 transfer effects are diverse and suggest that they are diverse for two reasons: variability in the information value of relevant phonetic cues in the L1, as well as variability in the degree to which linguistic expectations associated with a target L2 (or the lack thereof) predispose the listener to make effective use of these cues.

4aSC21. Steady as /ʃi/ goes: The spectral kinematics of sibilant fricatives in English and Japanese. Patrick F. Reidy and Mary E. Beckman (Dept. of Linguist, The Ohio State Univ., 24A Oxley Hall, 1712 Neil Ave., Columbus, OH 43210, patrick.francis.reidy@gmail.com)

Sibilant fricatives are often treated as having steady-state articulatory targets, which fix their spectra throughout their duration; however, Iskarous *et al.* (2011) reported that the centroid frequency of English /s/ varies considerably across the fricative's time course. This study replicates their spectral analysis using a psychoacoustic measure (peak ERB) and then extends it to English /ʃ/ and Japanese /s, ɕ/. The time-varying spectral pattern of each fricative was approximated with a nine-point peak ERB trajectory, computed from 20-ms windows spaced evenly throughout each token. There were three notable results. First, adults did not produce the same spectral kinematic pattern for all sibilants in a given language: the spectral peak of English /s/ followed a concave trajectory, while /ʃ/ remained relatively flat. Second, phonetically similar fricatives from different languages did not necessarily show similar dynamical spectral patterns: the peak trajectory of /s/ was curved in both languages, but reached its maximum much earlier in Japanese. Finally, three- to five-year-old children exhibited a developmental path toward language- and consonant-specific spectral patterns: as age increased, English-acquiring children produced /ʃ/ with decreasing curvature to its spectral peak trajectory, approaching that produced by the adults.

4aSC22. Effects of experience on the processing of phonetic contrasts in foreign-accented Spanish. Fernando Llanos (School of Lang. & Cultures, Purdue Univ., West Lafayette, IN 47907, fllanos@purdue.edu) and Alexander L. Francis (Speech, Lang. & Hearing Sci., Purdue Univ., West Lafayette, IN)

Non-native accented speech is typically less intelligible than unaccented speech. However, intelligibility improves with experience. Experience might improve intelligibility by guiding listeners' expectations regarding the systematic divergence of specific acoustic cues from the native norm within the non-native context. If an accent imposes predictable changes on the acoustic cue patterns present in speech, then listeners experienced with that accent may change their judgment of what was said based on whether or not it was perceived in an accented context. In the present study, two groups of native speakers of Spanish with and without significant experience with English-accented Spanish listened to Spanish sentences produced with and without a strong English accent. Each sentence ended in a Spanish word produced with or without English accent, but the voice onset time (VOT) of the first consonant in the word was artificially varied to form a continuum from bata (robe) to pata (paw). Experienced listeners showed a category boundary at a VOT of approximately 5 ms with no significant difference between accent conditions, suggesting that listeners were not affected by the perception of a familiar foreign accent. Additional results from inexperienced listeners and using non-word targets and fully English context sentences will be discussed.

4aSC23. Multiple sources of information contribute to novel category formation. Emily Cibelli (Linguist, UC Berkeley, 1203 Dwinelle Hall, Berkeley, CA 94720, ecibelli@berkeley.edu)

The acquisition of novel phonemes in a new language often presents a challenge for learners, particularly when target categories overlap with or assimilate to native categories. In the current study, English speakers who have no experience with Hindi are asked to learn the Hindi dental-retroflex place contrast and the four-way stop voicing contrast. The multi-day study includes an AX discrimination task and a repetition task with (V)CV syllables. In experiment 1, a training procedure was designed to manipulate multiple sources of information available to the listener. Training sessions include performance feedback and adaptive fading (a progression in exposure from clear tokens to more peripheral exemplars). Critically, the study also includes explicit articulatory training of the target sounds, to test the hypothesis that information about the existence of a new articulatory target can support the development of a perceptual category. Experiment 2, a control study, tests whether simple exposure to stimuli without feedback or training has any effect on performance. Without training, discrimination remained at baseline levels, suggesting that short-term exposure alone is insufficient for category formation. The place contrast presented the biggest challenge for listeners, with some improvement in performance if a voicing cue co-occurred with the place contrast.

4aSC24. Comparison across languages using the multilingual Matrix Test: Which language is best to survive in a cocktail party? Birger Kollmeier, Sabine Hochmuth, Tim Jürgens, Ania Warzybok, and Thomas Brand (Cluster of Excellence Hearing4all & Medizinische Physik, Universität Oldenburg, Cluster of Excellence Hearing4all, HörTech gGmbH, Oldenburg D-26111, Germany, birger.kollmeier@uni-oldenburg.de)

The Matrix test (i.e., sentence test with fixed syntactic structure, but ten alternative words in each position which may lead to nonsense utterances) has the potential to overcome the inherently language-dependent incompatibilities of speech audiology. It is meanwhile available (with varying degree of supportive data) in Swedish, German, Danish, Dutch, American English, British English, French, Polish, Turkish, Spanish, Italian, Persian, Arabian, Finnish, and Russian. Several measures have been taken to make the tests as efficient, reliable and comparable across different languages as possible and to establish a *de-facto* standard. Using the Matrix concept it is also possible to estimate the "communication efficiency" of the different languages for this kind of sentences against each other. To eliminate the influence of the individual speaker, recordings with accent-free bilingual speakers (German-Russian and German-Spanish) were used to assess the respective speech reception threshold (SRT) for native listeners using stationary and fluctuating background noise. The results show both an inter-speaker and inter-language effect in the order of 3 dB. The latter is larger between German and Spanish than between German and Russian. The origin of these effects (such as long-term speech spectrum and the relative information content of consonants and vowels) will be discussed.

4aSC25. Nasals resonances in diphthongization: A preliminary study by nasal and oral acoustic output recording separately. Rita Demasi and Didier Demolin (Gipsa-Lab, Université Stendhal , 1180, Ave. Centrale BP25, Grenoble, Rhone-Alpes 38031, France, ritademasi@gmail.com)

Our goal is to find new acoustical evidences that characterize the nasality correlation by recording the month and nostril signal separately. This allows visualize the nasal and vocalic resonances individually. Few studies cover the nasal diphthongization and this phenomenon combines a partial nasalized vowel and a nasal glide. In Brazilian Portuguese (BP) the diphthong /aw/ and /ãw̃/ are distinctive. All data were recorded by Handle Separator from Glottal Enterprises. This records two acoustics output in different channels using two microphones. The plate is supported between the mouth and the nose. Six speakers from Paulistano dialect were recorded. The corpus covered back diphthongs in offset: [paw]; [saw]; [maw]; [taw]; [kaw]; [pãw̃]; [sãw̃]; [mãw̃]; [kãw̃] and [tãw̃]. In this task, each subject had to read the carry-sentence three times: [dʒigʊ_todʊ dʒigɐ]. Because the acoustic outputs are mixed, this method simplifies the separation between the both signal. The partial results show that the waveform and the spectrogram in this signals have a different configuration. In /ãw̃/, the nasal waveform starts and finishes with a very-low energy. The higher energy is concentrated between the boundary of the both vocalic segments. In the spectrogram, the formants configuration are plates and they used to lose energy gradually (average: Fn1 374Hz, Fn2 2333 Hz, and Fn3 3049 Hz). This is different in oral format configuration where F2 has a descent movement and F3 has a ascend movement.

4aSC26. Effect of musical experience on tonal language perception. Abigail Chua and Jason Brunt (Biola Univ., 13800 Biola Ave, La Mirada, CA 90639, abigail.j.chua@biola.edu)

Potential connections between musical experience and language learning ability have been discussed and debated about in neurological and musical psychology literature. The identification of Mandarin tones was tested in non-Mandarin speakers. The dependent variable was the accuracy of tone identification in the mandarin phrases. A simple questionnaire was used to measure musical experience. Musical experience and experimental trial accuracy were related. Non-significant effects are also discussed. A t-test was used to compare identification accuracy of those with musical experience to those without musical experience. Musicians had higher test trial accuracy scores ($M = 0.311$, $SD = 0.056$), than nonmusicians ($M = 0.275$, $SD = 0.03$). The difference was significant $t(31) = 2.228$, $p = 0.033$, suggesting that the presence of musical training increases the effectiveness of skill in Mandarin tone identification for non-Mandarin speakers. There was a strong correlation for accuracy and overall years of practice $r(29) = 0.4$, $p = 0.02$. The more years of overall practice musicians had the higher the accuracy they demonstrated in the test trials. There was no correlation between accuracy and years of interval training, years of playing an instrument, years of advanced musical study, or years of current practice.

4aSC27. Acoustic variability in the speech of second language learners of American English as a function of accentedness. Bruce L. Smith (Communications Sci. and Disord., Univ. of Utah, 390 S. 1530 E., Rm. 1216, Salt Lake City, UT 84112, bruce.smith@hsc.utah.edu) and Rachel Hayes-Harb (Linguist, Univ. of Utah, Salt Lake City, UT)

The primary issue of interest in the present study concerned acoustic variability among L2 learners of English with different degrees of accentedness. Specifically, we were interested in determining whether L2 learners with stronger accents differ from L2 learners with weaker accents in terms of the amount of within-subject variability they manifest when producing English consonants and vowels. Twenty L2 English learners from nine different L1 backgrounds and a group of 20 native English control subjects produced a number of target sounds contained within CVC words that were embedded in a carrier phrase. Accent ratings for the twenty L2 talkers were obtained, and acoustic measurements were made of various consonants and vowels; coefficient of variation [(S.D. $\div$ mean) $\times$ 100] was also computed for each of the acoustic measures. A number of temporal and spectral comparisons were made between L2 talkers with stronger versus weaker accents and with the native control subjects. Results indicated that although L2 subjects with stronger accents sometimes showed greater inter-subject (i.e., group) variability, they did not typically show more within-subject (i.e., token-to-token) variability than subjects with weaker accents, regardless of how accurate they were in producing native-like consonants and vowels.

4aSC28. Native English speakers' perception of Arabic emphatic contrasts. Kristie Durham (Dept. of Linguist, Univ. of Utah, 255 S Central Campus Dr., Rm. 01400, Salt Lake City, UT 84112, Kristie.Durham@utah.edu), Aleksandra Zaba (Second Lang. Teaching and Res. Ctr., Univ. of Utah, Salt Lake City, UT), and Rachel Hayes-Harb (Dept. of Linguist, Univ. of Utah, Salt Lake City, UT)

In Arabic, emphasis (secondary velar/pharyngeal constriction) distinguishes some consonants. Native Jordanian Arabic speakers have been shown to rely more heavily on the rime than the onset of CVC syllables when identifying plain versus emphatic onsets (Jongman *et al.* 2011). We investigated whether native English speakers similarly rely on the rime when discriminating Arabic plain-emphatic pairs. We also investigated the influence of vowel quality on discrimination performance. Native English speakers (no Arabic experience) performed an AXB task involving cross-spliced CVCs with plain/emphatic onsets/rimes. Our subjects also relied more heavily on the rime than on the onset; this effect was most robust when the V was /æ/, followed by /u/ and /i/. A cross-language vowel identification task revealed that subjects identified Arabic /æ/ in emphatic contexts as systematically different English vowels than in plain contexts, with only 10% overlap in vowels identified. The overlap for /i/ and /u/ was much higher, at 84% and 91%, respectively. We thus found that native English listeners, like native Arabic listeners, rely on the rime to make onset emphasis judgments, this effect is moderated by vowel, and the influence of the preceding vowel may be related to the mapping between vowel allophones and English vowel categories.

4aSC29. Spontaneous speech variability across languages: Labial and velar stops. Natasha L. Warner, Miguel Simonet (Dept. of Linguist, Univ. of Arizona, Box 210028, Tucson, AZ 85721-0028, nwarner@u.arizona.edu), Benjamin V. Tucker (Linguist, Univ. of AB, Edmonton, AB, Canada), Dan Brenner, Maureen Hoffmann, Alejandra Baltazar, Andrea Morales, and Yamile Diaz (Linguist, Univ. of Arizona, Tucson, AZ)

Stops such as /ptkbdg/ are perhaps the most-studied type of consonant in all of phonetics, and they have well-defined acoustic properties that one expects to find in a typical pronunciation. However, casual spontaneous speech reveals highly variable realizations of stops, ranging from voiceless stops with silent closure, burst, and aspiration noise, to weak approximants with only a slight weakening of formants, to deletion. Even careful speech reveals considerable variability. We examine acoustic realizations of intervocalic stops in Dutch, Spanish, Japanese, and English, as well as the L2 English speech of the native Dutch, Spanish, and Japanese speakers. For each speaker, we measure data from spontaneous casual conversation and from careful word-list reading. In this presentation, we focus on realizations of /pbkg/. Preliminary results indicate that Dutch speakers variably transfer word-final devoicing of Dutch voiced stops to their English /bg/, but that they do not rely on the Dutch /x/ (orthographic "g") as a source of their English /g/. Results also show that Spanish /bg/ in conversation are almost categorically approximants or nearly deleted. Spanish speakers, especially those who learned English late, appear to apply this pronunciation to English consonants as a reduced speech style.

4aSC30. Prevention of learning of a non-native phonetic contrast by prior exposure to the contrasting stimuli while performing an irrelevant visual task. Beverly A. Wright, Jessica S. Conderman, Matthew K. Waggenspack (Commun. Sci. and Disord., Northwestern Univ., 2240 Campus Dr., Evanston, IL 60202, b-wright@northwestern.edu), and Nicole Marrone (Speech, Lang., and Hearing Sci., Univ. of Arizona, Tucson, AZ)

Exposure to an acoustic stimulus can facilitate learning when encountered near in time to practice on a perceptual task. Here, we explored the possibility that this learning enhancement arises in part because the stimulus exposures encountered in the absence of practice amplify the internal representation of the stimulus and this amplification then remains during the subsequent practice period. If this is the case, learning should be disrupted if the stimulus representation is instead suppressed during stimulus exposure without practice, because that suppression should also spread to the practice period. To test this idea, we trained listeners on a non-native phonetic-contrast categorization task using regimens in which a period of practice followed a period of stimulus exposure without practice in each daily session. We manipulated the extent to which listeners presumably suppressed the auditory stimuli that were presented without accompanying practice by varying the attentional demands of a visual task performed during their presentation. Learning decreased markedly as the attentional demand during these periods increased. Thus, it appears that the magnitude of the internal stimulus representation affects learning and that changes in this magnitude can spread beyond the time in which they are induced to promote or interfere with learning. [Work supported by NIH.]

4aSC31. The rhythm of Aviation English by Native American English speakers. Julia Trippe and Eric Pederson (Linguist, Univ. of Oregon, 439 Almaden St., Eugene, OR 97402, trippe@uoregon.edu)

Air traffic controllers (ATC) and pilots at international airports must speak Aviation English (AE). Native and non-native English speakers alike must learn and effectively communicate using this technical language based on standard English. This project calculates the rhythmic profile of Native Speaker Aviation English (NSAE), which serves as the target for learners of AE and against which potential communication failures can be evaluated. NSAE rhythmic profile can be contrasted with the first language (L1) prosody to evaluate learner AE production and model training methods for specific L1 AE learners. NSAE generally exhibits flat intonational contours, so we focus on rhythm metrics. Our previous study's findings demonstrated that NSAE metrics pattern differently than standard American English, falling between "stress-timed" and "syllable-timed" languages. Rhythm metrics based on consonant and vowel duration are affected by AE's lack of function words (i.e., fewer reducible vowels), standard phraseology (producing prosodic chunking), and rapid speech rate (reflecting compressibility differential between vowel and consonant segments). We are training an automatic speech aligner to segment ATC NSAE and calculating a baseline for American NSAE using qualitative metrics (Ramus 2000; Low *et al.*, 2000; Dellwo 2006). We will present our findings on how NSAE patterns with similarly evaluated languages.

4aSC32. The potential segmental influence of Taishanese (first language) on English (second language) intelligibility. Tracy Mai and Emily Wang (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 1611 West Harrison, Suite 530, Chicago, IL 60612, emily_wang@rush.edu)

This study examined the influence of segmental differences in Taishanese Chinese on English speech intelligibility. There are over half a million Taishanese speakers in the United States. However, very little is known about the linguistic interference between the two languages. We hypothesized that the English speech intelligibility of Late Bilingual speakers would be significantly reduced by the interference of their first language of Taishanese. Furthermore, the main source of their reduced speech intelligibility would be the segmental interference from the Taishanese on English. Speech data from a focused set of vowels and consonants as well as controlled spontaneous speech were collected from three different Speaker Types: Late Bilinguals (4), Sequential Bilinguals (2), and Monolingual English speakers (2). Acoustic analyses and perceptual experiment were conducted. The primary outcome measures were perceived speech intelligibility and mean Number of Real Words (NRW) per utterance. The secondary outcome measures were duration and formant frequencies of vowels, VOT for syllable-initial stops, and syllable duration of syllable-final stops. The results showed that the Late Bilingual speakers had significantly reduced English speech intelligibility ($p < 0.01$). The segmental-level differences from both vowels and consonants between Taishanese and English were responsible for the reduced speech intelligibility.

4aSC33. Perception of conversational and clear speech syllables by native and non-native English-speaking listeners. Catherine L. Rogers, Marissa Voors, and Jenna Luque (Dept. of Commun. Sci. and Disord., Univ. of South Florida, 4202 E. Fowler Ave., PCD1017, Tampa, FL 33620, crogers2@usf.edu)

In a recent study later, but not earlier, learners of English as a second language produced a smaller clear-speech benefit than native English-speaking talkers for vowels produced in six /bVd/ syllables (Rogers *et al.*, 2010, JASA **123**, 410–423). The present study compares perception of the same syllables by native and non-native English-speaking listeners. Conversational and clear-speech productions of the target syllables, "bead, bid, bayed, bed, bad," and "bod," were selected from three monolingual English speakers who had produced a significant clear-speech benefit in Rogers *et al.* (2010). The syllables were then mixed with noise at several signal-to-noise ratios (SNRs). Perception of these stimuli by three groups of listeners will be examined: (1) monolingual native English speakers, (2) 'early' learners of English as a second language, with an age of immersion (AOI) of 12 or earlier, and (3) later learners of English as a second language, with an AOI of 15 or later. Analyses of results of the six-alternative forced-choice task will focus on comparisons across listener groups, for the following measures: (1) estimates of clear-speech benefit at approximately 50% correct; (2) performance at a common SNR; and (3) estimates of the slope of the psychometric function. [Work supported by NIH.]

4aSC34. A virtual environment for modeling the acquisition of vowel normalization. Andrew R. Plummer (Ohio State Univ., 1712 Neil Ave, Columbus, OH 43210, plummer@ling.ohio-state.edu)

Vowel normalization is a computation that is meant to account for the differences in the absolute direct (physical or psychophysical) representations of qualitatively equivalent vowel productions that arise due to differences in speaker properties such as body size types, age, gender, and other socially interpreted categories that are based on natural variation in vocal tract size and shape. We present a virtual environment for vocal learning which provides the means to model the acquisition of vowel normalization, along with other aspects of vocal learning. The environment consists of models of caretaker agents representing five different language communities—American English, Cantonese, Greek, Japanese, and Korean—

derived from vowel category perception experiments (Munson *et al.*, 2010, Plummer *et al.*, 2013) and models of infant agents (Plummer, 2012, 2013) that "vocally interact" with their caretakers. Moreover, we develop a model of caretaker social and vocal signaling in response to infant vowel productions, and of an infant's internalization of these signals and the internal computations over them. More broadly, we model the acquisition of vowel normalization within a developmental framework encompassing a suite of vocal learning phenomena, including language-specific caretaker vocal exchanges, perceptual warping, and multisensory matching and narrowing.

THURSDAY MORNING, 8 MAY 2014 552 A/B, 8:00 A.M. TO 12:00 NOON

Session 4aSP

Signal Processing in Acoustics and Underwater Acoustics: Sensor Array Signal Processing I

Kainam T. Wong, Chair
Dept. of Electronic & Information Eng., Hong Kong Polytechnic Univ., DE 605, Hung Hom KLN, Hong Kong

Invited Papers

8:00

4aSP1. Deconvolution-based acoustic source localization and separation algorithms. Mingsian R. Bai and Chia-Hao Kuo (Power Mech. Eng., Tsing Hua Univ., 101 Sec. 2, Kuang_Fu Rd., Hsinchu 30013, Taiwan, msbai63@gmail.com)

In this paper, localization and separation of acoustic sources are examined. Depending on the number of sources in relation to the array channels, the problem is investigated in terms of underdetermined and overdetermined configurations. In the underdetermined configuration, virtual monopole sources are assumed in uniformly spaced angles. The problem is then formulated into compressive sampling (CS) problem which can be solved by using the linearly constrained -norm convex (CVX) optimization. The solution yields the directions of real sources and the source signal spectrum, which enables localization and reconstruction of sources at one shot. In the underdetermined configuration, source localization and signal separation is carried out in two steps. First, the directions of arrival (DOA) are estimated with Minimum Variance Distortionless Response (MVDR) or Multiple Signal Classification (MUSIC). Next, Tikhonov regularization (TIKR) is utilized to recover the source spectrum. In the localization problem for both configurations, Neyman-Pearson detector is employed to determine thresholds for source detection. Numerical and experimental results show that the proposed methods produce improved speech quality in terms of mean opinion score (MOS) in perceptual evaluation of speech quality (PESQ) test.

8:20

4aSP2. Reproduction of higher order virtual sources using loudspeaker arrays. Jung-Woo Choi and Yang-Hann Kim (Mech. Eng., KAIST, YuseongGu GuseongDong 373-1, Daejeon 373-1, South Korea, khepera@kaist.ac.kr)

Reproduction of a monopole virtual source has been extensively studied for providing the locatedness or directional information to the listener. The monopole virtual source is often assumed to be omni-directional, and the driving signals of multiple loudspeakers are determined such that the uniform radiation pattern can be reproduced in space and time. The virtual source of which radiation pattern consists of higher order radiation pattern, however, can also be reproduced for the control of perceived stage width or focusing of acoustic energy. In this work, we introduce various applications involving the higher order radiation pattern. The magnitude or phase distribution rapidly changing in space and time is reproduced by an array of loudspeakers, and the limitation of the conventional reproduction technique based on the stationary phase approximation, e.g., wave field synthesis, is demonstrated. A single layer formula to eliminate the artifact of the conventional technique is addressed.

8:40

4aSP3. Distance perception in the sound field reproduced by a linear loudspeaker array. Dong-Soo Kang, Jung-Woo Choi, and Yang-Hann Kim (Mech. Eng., KAIST, 291 Daehak-ro, Yuseong-gu, KAIST, ME Bldg Rm#4114, Daejeon 305701, South Korea, dooly0819@kaist.ac.kr)

When a sound field from a virtual sound source is reproduced by a linear loudspeaker array, a listener in the sound field can perceive the distance of the virtual source, as well as its direction. It has been known that the perception of distance is affected by many acoustic parameters, such as the loudness change, direct-to-early reflection energy ratio, and interaural level difference (ILD). Among them, ILD is the dominant cue to perceive distance when the source is located near the lateral side of the listener [Brungart and Rabinowitz, J. Acoust. Soc. Am. **106**(3), Sept. 1999]. Nevertheless, ILD of the reproduced sound field are not identical to that of the target sound field, because the loudspeaker array reproducing the sound field has many practical limitations such as spatial aliasing and truncation of array aperture. To identify these artifacts, especially for the virtual source in a close proximity to the listener, a head-scattering model is constructed using a simple rigid sphere. The ILDs at various head locations are then calculated and compared to those of the target sound field. From the observations on ILD change, a driving function is modified to reconstruct ILDs of the target sound field.

Contributed Papers

9:00

4aSP4. Broadband acoustic-source localization using passive sonar via multitask learning. Pedro A. Forero and Paul A. Baxley (Maritime Systems Div., SPAWAR Systems Ctr. - Pacific, 53560 Hull St., San Diego, CA 92152, forer002@umn.edu)

Passive sonar is an attractive technology for underwater acoustic-source localization that enables the localization system to conceal its presence and does not perturb the maritime environment. Notwithstanding its appeal, passive-sonar-based localization is challenging due to the complexities of underwater acoustic propagation. Different from alternatives based on matched-field processing whose localization performance severely deteriorate when localizing multiple sources and when faced with model mismatch, this work casts the broadband underwater acoustic-source localization problem as a multitask learning (MTL) problem, thereby enabling robust and high-resolution localization. Here, each task refers to a sparse signal approximation problem over a single frequency. MTL provides an elegant framework for exchanging information across the individual regression problems and constructing an aggregate (across frequencies) source localization map. The localization problem is formulated as a stochastic least-squares optimization problem with a group sparsity constraint enforcing a common support across frequency maps. Efficient algorithms based on block coordinate descent are developed for solving the localization problem. Predictor screening rules are also developed to further reduce the computational complexity of the proposed method. Numerical tests on real data illustrate and compare the localization performance of the proposed algorithm to that of competitive alternatives.

9:15

4aSP5. Transient detection via acoustic particle velocity multi-mission sensor. Latasha Solomon and Leng Sim (US Army Res. Lab, 2800 Powder Mill RD, Adelphi, MD 20783, latasha.i.solomon.civ@mail.mil)

In this research, we compare the direction of arrival (DOA) accuracy of a micro-electro-mechanical systems (MEMS) based acoustic particle velocity sensor developed by Microflown Technologies with that of a collocated, 1-m tetrahedral array. When deployed as an unattended sensor system, the Acoustic Multi-Mission Sensor (AMMS) greatly facilitates hardware set-up and periodic maintenance. An array of microphones is now replaced by a single sensor, saving in overall system cost, size, weight, and power usage. The single sensor has the capability to measure both the (scalar) sound pressure and the (vector) acoustic particle velocity, thus providing DOA estimates. This research will explore performance and determine limitation of the two sensors in complex environments as well as open fields for detection of both small arms fire (SAF) and rocket propelled grenades (RPGs).

9:30

4aSP6. A study of broadband sensor location selection using convex optimization in very large scale arrays. Yenming Lai and Radu V. Balan (Appl. Mathematics, Statistics, and Sci. Computation, Univ. of Maryland, 5010 Pierce Ave., College Park, MD 20740, yenming.mark.lai@gmail.com)

Consider a sensing system using a large number of N microphones, placed in multiple dimensions to monitor a broadband acoustic field. Using all the microphones at once is impractical because of the amount of data generated. Instead, we choose a subset of D microphones to be active. Specifically, we wish to find the set of D microphones which minimizes the energy of the interference gains at multiple frequencies while monitoring a target of interest. A direct, combinatorial approach—testing all N choose D subsets of microphones is impractical because of problem size. Instead, we use a convex optimization technique that induces sparsity through a l1-penalty to determine which subset of microphones to use. We measure the energy of the interference gains in three ways: the maximum gain, the average gain, and the average squared gain and compare the results. Furthermore, we assume one reflection off of each wall in our problem setup and minimize the gains of the reflections. We test the robustness of the our solution through simulated annealing and compare its performance against a classical beamformer which maximizes SNR. We also do exhaustive searches to compare the performance of our algorithm against the global optimum.

9:45

4aSP7. Estimation algorithm coordinates source signal towed long antenna. Igor Y. Anikin (Concern CSRI Elektropribor, JSC, 30, Malaya Posadskaya Str., St. Petersburg 197046, Russian Federation, anikin1952@bk.ru)

In some sonars is require the use of towed long antenna. By "large"long antenna is understood conventionally antenna directivity pattern width is less than some value, for example, less than 1°. Due to the large antenna length when towing a change in its form, as well as change of coordinates the source relative to antenna. For this reason, the source coordinates are determined with errors. The report discusses an algorithm for estimating the coordinates the source of towed long antenna. Algorithm consists in the separation of the antenna into several sections. Section length is chosen so that the antenna directivity pattern width of the section was 1°... 2°. Each section is formed by the fan of directivity patterns. The joint processing of signals from the output of the directivity patterns of the fan formed by sections of the antenna provides the coordinates of the source. Results of mathematical modeling error estimates of coordinates, offered algorithm are compared with the potential errors that follow from the Cramer-Rao inequality.

10:00–10:15 Break

10:15

4aSP8. Random matrix theory model for mean notch depth of the diagonally loaded minimum variance distortionless response beamformer for a single interferer case. Saurav R. Tuladhar, John R. Buck (ECE, Univ. of Massachusetts Dartmouth, 285 Old Westport Rd., ECE Dept, UmassD, North Dartmouth, MA 02747, stuladhar@umassd.edu), and Kathleen E. Wage (ECE, George Mason Univ., Fairfax, VA)

Adaptive beamformers (ABFs) suppress interferers by placing notches in the beampattern at interferer directions. This suppression improves the detection of weaker signals of interest even in the presence of strong interferers. The magnitude of the notch depth (ND) is an important parameter governing the adaptive gain obtained from using ABFs over conventional beamforming in the presence of interferers. This research derives models for the mean ND of a diagonally loaded minimum variance distortionless response (MVDR) beamformer for a single interferer case. The model describes the mean ND as a function of the number of snapshots, the number of sensors in the array, the interferer to noise ratio (INR) level, the interferer direction, and the diagonal loading level. The derivation exploits random matrix theory (RMT) results on the behavior of the eigenvectors of the spiked covariance matrix. The RMT based ND model predictions are in close agreement with simulation results over a range of INR values and number of snapshots.

10:30

4aSP9. Improved modal dispersion estimation using vertical array beamforming. Valerie Vinciullo (Appl. Physical Sci. Corp. , 4 Hillside Ave., unit 2, Pawcatuck, Rhode Island 06379, vvinciullo@my.uri.edu), Gopu R. Potty (Dept. of Ocean Eng., Univ. of Rhode Island, Narragansett, RI), Kevin Cockrell (Appl. Physical Sci. Corp. , Groton, CT), and James H. Miller (Appl. Physical Sci. Corp. , Narragansett, Rhode Island)

Geoacoustic inversions using modal dispersion data is a very robust technique to estimate properties of the shallow water sediments. Accurate estimation of the modal arrival times is required for improving the accuracy of the inversion. A time-frequency analysis of the single hydrophone data is typically used to extract modal arrival times. This study explores the possibility of incorporating the data from a vertical line array (VLA) to enhance the accuracy of arrival time estimation. The method relies on beam forming in horizontal wavenumber at each instant in time to produce a time-frequency-wavenumber diagram (movie) which will provide an extra dimension to help separate the modes. For a given time and horizontal wavenumber, the arrival time is unique since the group speed is uniquely determined by the frequency and horizontal wavenumber. So, even if the time and wavenumber resolution is not sufficient identify individual modes, the 3-D surface plot of arrival time versus frequency and wavenumber can be created. The shape of that surface can be compared to simulated surfaces for geoacoustic inversion, even if mode arrivals appear to overlap. This approach will be tested using synthetic data and the length and spacing requirements of the VLA will also be investigated. [Work supported by Office of Naval Research, code 322OA.]

10:45

4aSP10. Matched-field source localization with non-synchronized sensor arrays. Stan E. Dosso (School of Earth & Ocean Sci, Univ of Victoria, PO Box 1700, Victoria, BC V8W 3P6, Canada, sdosso@uvic.ca)

This paper considers matched-field ocean acoustic source localization based on acoustic field measurements at an array of sensors which are not synchronized in time or at an array which is comprised of non-synchronized combinations of synchronized sub-arrays. Standard matched-field methods are based on acoustic-field measurements at a time-synchronized array to allow coherent processing over space (the array aperture). For non-synchronized systems, frequency-coherent/space-incoherent processing can be applied if the complex source spectrum (amplitude and phase) is known, but this is rarely the case in practical applications. However, time- and frequency-coherent processing are not the only possibilities. Maximum-likelihood methods can be applied to derive optimal matched-field processors for any state of source/receiver information. Using this method, optimal processors can be developed for broadband matched-field localization with any combination of synchronized and/or non-synchronized components based on the fact that the source amplitude spectrum is the same (although unknown) for all receivers (the phase spectrum is both unknown and variable for non-synchronized components). Bayesian inversion methods are employed to quantify the source-localization information content for various array scenarios.

11:00

4aSP11. Research on rotary spiral array applied in near-field acoustical holography. Chen Lin-Song (Power Eng. Dept., Naval Univ. of Eng., Jiefang St. 717, Wuhan, Hubei 430033, China, 13294153193@163.com)

This paper presents a new method to apply a spiral array in nearfield acoustical holography (NAH). Usually, a NAH array needs much more microphones than beamforming array does. Superior to a uniform planar array or linear scanning array, this spiral array rotates to get more measuring data. Without any static referring microphone, a numerical method was suggested to estimate the phase difference measured at different time. Numerical simulations and a series experiment confirmed that this method is adequate for the sound below 450 Hz. It is especially useful for using a random planar array at NAH mode, while the beamforming mode can only cover the higher frequency band.

11:15

4aSP12. A constrained adaptive beamforming algorithm for spherical microphone arrays. Gary W. Elko (mh Acoust. LLC, 25A Summit Ave., Summit, NJ 07901, gwe@mhacoustics.com) and Jens M. Meyer (mh Acoust., Fairfax, Vermont)

In this presentation, we will present a novel constrained adaptive beamformer algorithm that utilizes an inherent property of spherical harmonic eigenbeams which form the bases signals for spherical microphone array beamforming. Two simple constraints are placed on the weights to preclude the adaptive beamformer from nulling signals arriving from a desired "look" direction. The adaptive algorithm has been simulated for some simple acoustic fields as well as a diffuse field. We have implemented the algorithm in realtime on mh acoustics em32 Eigenmike® spherical microphone array and we will present some measurement results.

11:30

4aSP13. A rigid-body model for diffraction imaging of solid objects: Theory and experimental results. Edward H. Pees (Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, edward.pees@navy.mil)

Acoustical imaging via the method of diffraction tomography is typically applied to weakly scattering, fluid objects, wherein the first Born approximation holds. Nonetheless, the technique can be applied to strong scatterers in a meaningful way if an appropriate object function is considered. In this talk, the theoretical form of the object function for a rigid body is developed along with an inversion formula for centripetal, broadband data collection. Applying the latter to experimental, underwater echo data from a variety of objects, reconstructions are presented and interpreted in terms of a Kirchhoff boundary condition. The approach can potentially reveal the relative importance of different scattering mechanisms in the overall pressure field reflected from a body by how closely the rigid body object function is reconstructed. Morphological characteristics may also be identified for objects that are, for example, hidden or buried.

11:45

4aSP14. Multichannel myopic deconvolution using ambient noise sources. Ning Tian, Justin Romberg (School of Elec. and Comput. Eng., Georgia Inst. of Technol., 30 5th St. NE, Unit 606, Atlanta, GA 30308, ningtian@gatech.edu), and Karim Sabra (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA)

The ocean ambient noise has been increasingly utilized for ocean passive sensing and monitoring applications. By recording the received signals from the same individual noise source (for example, the shipping noise) at multiple hydrophones simultaneously, we develop a framework, called multichannel myopic deconvolution, which can allow us jointly estimate the source and the channel responses without any assumption about the source, but using some priori knowledge of the channel. Our work on this classical signal processing problem has two novel aspects. First, we recast the corresponding bilinear system of equations as a linear system with a rank constraint. This allows us to apply recently developed algorithms and analytical tools from the field of low-rank recovery to the blind channel estimation problem, yielding insight into the conditions under which accurate channel estimation is possible. Second, we incorporate (continuous-time) parametric uncertainty about the Green's functions as subspace constraints in the low-rank recovery problem. These subspaces are generated in a systematic way using the singular value decomposition, and their dimension can be directly related to the amount of priori knowledge we have about the channel. We will present simulations in shallow water environments of the proposed approach from relatively short observation times.

Session 4aUW

Underwater Acoustics: Acoustic Vector Sensor Measurements: Basic Properties of the Intensity Vector Field and Applications I

David R. Dall'Osto, Cochair
Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105

Peter H. Dahl, Cochair
Appl. Phys. Lab., Univ. of Washington, Mech. Eng., 1013 NE 40th St., Seattle, WA 98105

Chair's Introduction—8:20

Invited Papers

8:25

4aUW1. Using hydrophones as vector sensors. Selda Yildiz, LeRoy M. Dorman, W. A. Kuperman (Scripps Inst. of Oceanogr., University of California, San Diego, La Jolla, CA 92093-0238, wkuperman@ucsd.edu), Karim Sabra (School of Mech. Eng., Georgia Inst. of Tech, Atlanta, GA), Philippe Roux (Institut des Sci. de la Terre, Universite Joseph Fourier, Grenoble, France), Dale Green (Teledyne Benthos, 49 Edgerton Dr, N. Falmouth, MA), Stephanie Fried, and Henrik Schmidt (Mech. Eng., Mass. Inst. of Tech., Cambridge, MA)

Hydrophone arrays with spacing much less than an acoustic wavelength can be converted to vector sensors. Subsequent vector sensor signal processing can then be applied. Two particular applications are presented: The first is converting very low frequency acoustic data to seismic type data that contain polarization information and the second is getting directional information from sub wavelength acoustic arrays. We start with a review of the simple theory followed by some illustrative simulation examples. We then apply these signal processing methods to ocean acoustic data.

8:45

4aUW2. Tank acoustics, and sound source localization by plainfin midshipman fish (*Porichthys notatus*). David Zeddies (JASCO Appl. Sci., 2004 Coleridge Dr., #101, Silver Spring, MD 20902, David.Zeddies@jasco.com), Michael D. Gray, Peter H. Rogers (George W. Woodruff School of Mech. Eng., Georgia Inst. of Technol., Atlanta, GA), Richard R. Fay (Parmly Hearing Inst., Loyola Univ., Falmouth, Massachusetts), and Joseph A. Sisneros (Dept. of Psych., Univ. of Washington, Seattle, WA)

A series of experiments was undertaken to investigate methods of sound source localization by fish. In these experiments, positive phonotaxic responses of gravid female plainfin midshipman fish (*Porichthys notatus*) to low-frequency, playback tones (80–90 Hz) were studied as they approached sound sources. The sound fields for simple (monopole) and relatively complex (dipole) sources within the behavioral arena were measured and characterized in terms of pressure and particle motion. Results indicate that female midshipman fish are able to locate sound sources in the near field using acoustic cues alone, and that they used the particle motion vectors to locate the source in both the monopole and dipole sound fields. The tank acoustics were modeled and compared to the measured pressure and particle motion sound fields. [This work was supported by the National Science Foundation.]

Contributed Papers

9:05

4aUW3. Real-time acoustic monitoring and source level estimates of walruses in the northeastern Chukchi Sea using particle velocity sensors. Xavier Mouy (JASCO Appl. Sci., Victoria, Br. Columbia, Canada), Julien Delarue, Bruce Martin (JASCO Appl. Sci., 32 Troop Ave., Ste. 202', Dartmouth, NS B3B 1Z1, Canada, bruce.martin@jasco.com), David Hannay (JASCO Appl. Sci., Victoria, Br. Columbia, Canada), Chadwick Jay, and Anthony Fishbach (US Geological Survey, Anchorage, AK)

Particle motion sensors measure the vector component of the sound field. In underwater acoustics, they are used for studying the physics of the sound field, evaluating the potential effects of sound on fish, and defining the direction of arrival (DOA) of sound sources. Measuring the DOA in the vertical and horizontal plane allows two separate receivers to localize an acoustic source in three dimensions. In July 2013, we used two custom-built, real-time particle velocity acoustic recording systems to record and localize vocally active walruses in the water near groups hauled out on ice in the northeastern Chukchi Sea. The system was equipped with a three-axis dipole sensors and a calibrated omni-directional hydrophone. It was deployed at the water surface and transmitted data in real-time to a support skiff. The range between the recorders, support skiff, and calling animals was usually less than 200 m and typically within a few tens of meters, allowing for simultaneous visual observations. Calling walruses were localized using cross-fixes of acoustic bearings. Source levels were estimated by adding modeled frequency-dependent transmission losses to the received levels in each 1/3-octave-band obtained from the calibrated omni-directional hydrophone. Only calls with high signal-to-noise ratio were used in this analysis. The use of the particle velocity sensor allowed for the first source level measurements of walrus grunts and bell calls in the wild.

9:20

4aUW4. Estimates of the bottom reflection coefficient involving vector sensor. Jee Woong Choi (Dept. of Marine Sci. & Convergent Technol., Hanyang Univ., 55 Hanyangdaehak-ro, Ansan 426-791, Korea, Republic of, choijw@hanyang.ac.kr), David R. Dall'Osto, and Peter H. Dahl (Appl. Phys. Lab. and Mech. Eng. Dept., Univ. of Washington, Seattle, WA)

Estimates of the bottom reflection coefficient made in the frequency range of 4–8 kHz, as part of the Targets Reverberation Experiment (TREX) are presented. The TREX experiment took place in the Gulf of Mexico, near Panama City, FL, in waters 20 m deep. At the measurement site the sediments are loosely classified as fine sand. The reflection coefficient R is estimated over the nominal grazing angular range 1 to 20 deg., using measurements made at ranges 50 to 800 m and received on a vertical line array (length 1.6 m). The arrival time and magnitude of the bottom reflection is determined by the matched filtered output of a frequency modulated signal, 4–8 kHz. In addition, the match filter processing technique is applied to the vector sensor data (measured simultaneously and co-located with the line array.) This allows for an extraction of the active intensity contribution associated with the bottom reflection, and provides a vector intensity-based estimate of the bottom reflection coefficient. The estimates of the bottom reflection arrival time are also used to time-gate simultaneously transmitted cw tones (1–4 kHz) to analyze the Lloyd's mirror pattern associated with seabed reflection. [Research supported by ONR, with partial support from ONRG.]

9:35

4aUW5. Underwater techniques to characterize the near scattered acoustic vector field. Robert J. Barton, Geoffrey R. Moss, Brian K. Amaral, Georges Dossot (NUWC, 1176 Howell St., Bldg. 1320, Code 1524, Rm. 260, Newport, RI 02841, georges.dossot@navy.mil), and Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA)

In this study, we investigate the properties of the scattered acoustic vector fields generated by simple geometric objects, including the infinite rigid plate, disk, and sphere. Analytical solutions are derived from acoustic target strength scattering models in the near-field region. Of particular interest is the understanding of the characteristics of energy flow of the scattered acoustic vector field in the near- to far- field transition region. We utilize the time and space separable instantaneous active and reactive acoustic intensities to investigate the relative phase properties of the scattered field. Numerical results are presented for the near region scattered acoustic vector field of simple objects in both two and three dimensions. Previous in-air measurements are summarized, and an approach to taking water-borne measurements is offered.

9:50

4aUW6. The interference structure of very low frequency vector acoustic field and its positioning application. Sun Dajun, Shi Junjie, Lv Yunfei, Lan Hualin, and Mei Jidan (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ. and College of Underwater Acoust. Eng., Harbin Eng. Univ., Rm. 1005, Shuisheng Bldg., No.145 Nantong St., Nangang District, Harbin, Heilongjiang, China, Harbin 150001, China, junjieshi@hrbeu.edu.cn)

Vector hydrophone has natural dipole beam-pattern, and can also simultaneously and colocatedly measure the scalar and vector information of ocean acoustic field, which makes it convenient to determine the direction of arrival (DOA) and represent the stable interference structure of vector acoustic field. Utilizing the DOA and interference structure information together acquired by single vector hydrophone, it is able to position the target of interest. Firstly, bearing-time course for passing-by target is obtained by using the line or continuous spectrum imbedded in the received signal of vector hydrophone. Then, CPA ratio between the CPA range and speed of target as well as CPA instant are estimated on the basis on LMS criteria. Finally, the speed of target can be determined through the theoretically predicted interference range and the real interference time interval to fulfill target positioning including DOA and range. The idea was effectively validated during the experiment that took place in October of 2010 in South China Sea nearly 100m depth. Moreover, the idea presented here can be easily extended for further application such as combining the waveguide invariant. [Work supported by the National 863 Project (No. 2011AA090502) and National Defense Foundation Project (B2420132004).]

10:05–10:25 Break

Invited Paper

10:25

4aUW7. Acoustic energy streamlines in inhomogeneous fluids. Oleg A. Godin (CIRES, Univ. of Colorado and NOAA Earth System Res. Lab., Physical Sci. Div., Mail Code R/PSD99, 325 Broadway, Boulder, CO 80305-3328, oleg.godin@noaa.gov)

First introduced by Umov in 1873, wave energy streamlines offer an intuitive and informative description of energy flow much like conventional streamlines do for mass flow in fluid mechanics. Growing availability and increasing practical applications of acoustic vector sensors, such as sound-intensity meters, have led to a surge of interest to energy streamlines. In contrast to rays, which are essentially an asymptotic, short-wave concept, energy streamlines adequately represent arbitrary acoustic fields and reveal intricate and often unexpected details of the acoustic energy flow. Modern usages of the energy streamlines include studies of wave front dislocations, source localization, energy vortices in compressible fluids and elastic waveguides, and bounded beam diffraction. This paper will focus on applications of the energy streamlines to the description of reflection and refraction of acoustic waves at interfaces and to localization of low-frequency sound sources.

Contributed Papers

10:45

4aUW8. Observations of elliptical particle motion in shallow water and its dependence on source depth. David R. Dall'Osto (Acoust. Dept., Appl. Phys. Lab. Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Peter H. Dahl (Mech. Eng., Univ. of Washington, Seattle, WA)

Acoustic particle motion that follows an elliptical path coincides with a non-zero curl of the time-averaged intensity. This vector property is also observed as curvature in intensity (energy-flux) streamlines. Measurements of the acoustic intensity field made in shallow water are presented, along with simulations of the intensity field, to demonstrate some interesting relations between acoustic intensity and elliptical particle motion. Specifically, the direction in which intensity streamlines bend (sign of the curl of intensity) corresponds to the polarization of acoustic particle motion. For a source located in water, the polarization of particle motion depends on the modes of the underwater waveguide excited at a particular source depth. By raising a source up through the water column, an abrupt change in the polarization of particle motion can occur. This effect is examined with vector sensor data collected during an experiment near Panama City, FL. For a source located in air, elliptical particle motion is most evident a few wavelengths below the sea-surface where the contribution of the lateral (evanescent) wave is significant. This effect is examined with a recording of aircraft noise on both sides of the air-water interface made near Oak Harbor, WA.

11:00

4aUW9. Modeling the acoustic vector field to simulate glider-based acoustic processing methods. Georges Dossot (NUWC, 1176 Howell St., Bldg. 1320, Code 1524, Rm. 260, Newport, RI 02841, georges.dossot@navy.mil), Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA), and Edmund J. Sullivan (Prometheus Inc., Portsmouth, RI)

The feasibility of underwater gliders as passive acoustic receiving platforms is explored through simulated data. Over the last decade gliders have proven their worth as operational platforms to the oceanographic community, yet their merits as acoustic sensing platforms remain largely underexplored. Recently, the Office of Naval Research has equipped the Naval Postgraduate School with several gliders, which have now been fitted with acoustic vector sensors. To simulate real-world performance, the intensity vector field is modeled using the three-dimensional Cartesian version of the Monterey-Miami parabolic equation (MMPE) algorithm, which relies upon a split-step Fourier approach. Environmental information representative of the glider's sawtooth profile is incorporated as a three-dimensional sound speed profile, and incorporated into the PE model. These simulated data serve as the basis for signal processing techniques applicable to glider-based experimentation.

11:15

4aUW10. The influence of directional sea-surface waves on the acoustic intensity vector field. David R. Dall'Osto (Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105, dallosto@apl.washington.edu) and Peter H. Dahl (Mech. Eng., Univ. of Washington, Seattle, WA)

The effects of a rough sea surface on shallow water acoustic propagation are examined using experimental data collected from the Target and Reverberation Experiment (TREX) which took place off of the coast of Panama City, Florida in May 2013. During the experiment, the sea surface directional-wave spectrum was measured by a pair directional buoys moored at the experimental site. Acoustic measurements were collected using a bottom deployed recording tower (depth 20 m), that coherently recorded data from an accelerometer-based vector sensor, and a horizontal and vertical line array. Measurements using an active source, lowered from the stern of a research vessel, were made along propagation paths perpendicular and parallel to the surface wind-waves at source receiver ranges corresponding to approximately 10, 20, and 40 water depths. Results show that the directional properties of the rough sea-surface influence both the azimuthal and vertical distribution of the forward scattered intensity. A frequency dependence in vertical angular spreading is identified for the frequency range 1 to 3 kHz. A partial explanation for this effect originates from differences in the directional wave spectral level corresponding to forward scattering Bragg wavenumbers that are computed from the angles of the trapped modes.

11:30

4aUW11. Research on the double vector hydrophones' location for underwater low frequency source depth identification. Anbang Zhao, Xuejing Song, Bin Zhou, and Xuejie Bi (College of Underwater Acoust. Eng., Harbin Eng. Univ., Harbin Eng. University Underwater Acoust. Bldg., Rm. 812, Harbin, Heilongjiang, Harbin 150001, China, zhaoanbang@hrbeu.edu.cn)

The double vertically arranged vector hydrophones' pressure and horizontal velocity cross spectrum in Pekeris waveguide is derived, and the sign distribution of its active component is analyzed. The sign distribution varies with the horizontal ranges and source depths regularly, the signs change in a certain depth and the depth is defined as critical depth. By locating the vector hydrophones properly, a critical depth which is independent of horizontal range can be obtained, and this characteristic can be used for discriminating the source depth. The method of forecasting the vector hydrophones' locating depths according to the requirements of the critical depth is studied, and the forecast accuracy is validated by the simulation results. A reasonable set of the critical depth is conducive to discriminate the source depth accurately and effectively, which has extensive application prospects.

11:45

4aUW12. Joint estimation of frequency and azimuth using acoustic vector sensor signals based on sparse decomposition theory. Jinshan Fu (Sci. and Technol. on Underwater Acoust. Lab., Harbin Eng. Univ., Nantong St. 145, Haerbin 150001, China, fujinshan@hrbeu.edu.cn)

Acoustic vector sensor can obtain more information of sound field compared with scalar hydrophone. The sparse decomposition theory was put forward in the 90s of last century, and it provides a simple, flexible, and self-adaptive representation method of signal. Through sparse decomposition theory, it can essentially reduce the cost of signal processing and improve the compression efficiency. Space-time array manifold is constructed through signal analysis of single acoustic vector sensor (AVS). Based on sparse decomposition theory, the frequency and azimuth estimation algorithm is proposed, the frequencies and azimuths of multi-targets are estimated simultaneously by the joint estimation algorithm. Results using simulated data received from single acoustic vector sensor are illustrated. The accurate estimation of multi-targets' frequencies, azimuths, and signal amplitudes can be obtained using the estimation algorithm we deduced. Then, the influence of targets number, signal-to-noise (SNR), snapshots number on algorithm performance is analyzed.

Session 4pAA

Architectural Acoustics and Psychological and Physiological Acoustics: Psychoacoustics in Rooms I

Philip W. Robinson, Cochair
Media Technol., Aalto Univ., PL 15500, Aalto 00076, Finland

Frederick J. Gallun, Cochair
National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239

Chair's Introduction—1:00

Invited Papers

1:05

4pAA1. Introduction to "Psychoacoustics in Rooms," and tutorial on architectural acoustics for psychoacousticians. Philip W. Robinson (Media Technol., Aalto Univ., PL 15500, Aalto 00076, Finland, philrob22@gmail.com)

This special session—"Psychoacoustics in Rooms"—was born from the observation that psychoacoustics and room acoustics are often highly interleaved topics. Those researching the former attempt to determine how the hearing system processes sound, including sound from within specific environmental conditions. Practitioners of the latter aim to produce architectural enclosures catered to the auditory system's needs, to create the best listening experience. However, these two groups do not necessarily utilize a common vocabulary or research approach. This session, a continuation of one with the same name held at Acoustics 2012 Hong Kong, is intended to appeal to both types of researchers and bring them towards a common understanding. As such, the first two presentations are basic surveys of each paradigm. This presentation will focus on common architectural acoustic methods that may be of interest or utility to psychoacousticians.

1:25

4pAA2. A tutorial on psychoacoustical approaches relevant to listening in rooms. Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, Frederick.Gallun@va.gov)

From the year of its founding, members of the Acoustical Society of America have been interested in the question of how the acoustical effects of real-world environments influence the ability of human beings to process sound (Knudsen, "The hearing of speech in auditoriums," JASA **1**(1), 1929). While interest in this topic has been constant, the specialization of those focused on architectural acoustics and those focused on psychological and physiological acoustics has increased. Today, it is easily observed that we are likely to use methods and terminology that may be quite unfamiliar to those discussing a very similar question just down the hall. This presentation will survey a few of the most influential psychoacoustical approaches to the question of how the detection and identification of stimuli differs depending on whether the task is done in a real (or simulated) room as opposed to over headphones or in an anechoic chamber. The goal will be to set the stage for some of the talks to come and to begin a discussion about methods, terminology, and results that will help turn the diverse backgrounds of the participants into a shared resource rather than a barrier to understanding.

1:45

4pAA3. Speech intelligibility in rooms: An integrated model for temporal smearing, spatial unmasking, and binaural squelch. Thibaud Leclère, Mathieu Lavandier (LGCB, Université de Lyon - ENTPE, rue Maurice Audin, Vaulx-en-Velin, Rhônes 69518, France, thibaud.leclere@entpe.fr), and John F. Culling (School of Psych., Cardiff Univ., Cardiff, Wales, United Kingdom)

Speech intelligibility predictors based on room characteristics only consider the effects of temporal smearing of speech by room reflections and masking by diffuse ambient noise. In binaural listening conditions, a listener is able to separate target speech from interfering sounds. Lavandier and Culling (2010) proposed a model which incorporates this ability and its susceptibility to reverberation, but it neglects the temporal smearing of speech, so that prediction only holds for near-field targets. An extension of this model is presented here which accounts for both speech transmission and spatial unmasking, as well as binaural squelch in reverberant environments. The parameters of this integrated model were tested systematically by comparing the model predictions with speech reception thresholds measured in three experiments from the literature. The results showed a good correspondence between model predictions and experimental data for each experiment. The proposed model provides a unified interpretation of speech transmission, spatial unmasking, and binaural squelch.

2:05

4pAA4. Reverberation and noise pose challenges to speech recognition by cochlear implant users. Arlene C. Neuman (Dept. of Otolaryngol., New York Univ. School of Medicine, 550 First Ave., NBV 5E5, New York, NY 10016, arlene.neuman@nyumc.org)

The cochlear implant (CI) provides access to sound for a growing number of persons with hearing loss. Many CI users are quite successful in using the implant to understand speech in ideal listening conditions, but CI users also need to be able to communicate in noisy, reverberant environments. There is a growing body of research investigating how reverberation and noise affect speech recognition performance of children and adults who use cochlear implants. Findings from our own research and research from other groups will be reviewed and discussed.

2:25

4pAA5. Combined effects of amplitude compression and reverberation on speech modulations. Nirmal Kumar Srinivasan, Frederick J. Gallun (National Ctr. for Rehabilitative Auditory Res., 3710 SW US Veterans Hospital Rd., Portland, OR 97239, srinivan@ohsu.edu), Paul N. Reinhart, and Pamela E. Souza (Northwestern Univ. and Knowles Hearing Ctr., Evanston, IL)

It is well documented that reverberation in listening environments is common, and that reverberation reduces speech intelligibility for hearing impaired listeners. it has been proposed that multichannel wide-dynamic range compression (mWDRC) in hearing aids can overcome this difficulty. However, the combined effect of reverberation and mWDRC on speech intelligibility has not been examined quantitatively. In this study, 16 nonsense syllables (/aCa/ format) recorded in a double-walled sound booth were distorted using virtual acoustic methods to simulate eight reverberant listening environments. Each signal was then run through a hearing-aid simulation which applied four-channel WDRC similar to that which might be applied in a wearable aid. Compression release time was varied between 12 and 1500 ms. Consonant confusion matrices were predicted analytically by comparing the similarity in the modulation spectra for clean speech and compressed reverberant speech. Results of this acoustical analysis suggest that the consonant error patterns would be strongly influenced by the combination of compression and reverberation times. If confirmed behaviorally and extended to wearable hearing aids, this outcome could be used to determine the optimum compression time for improved speech intelligibility in reverberant environments. [Work supported by NIH R01 DC60014 and R01 DC011828.]

2:45–3:00 Break

3:00

4pAA6. Model of binaural speech intelligibility in rooms. Thomas Brand, Anna Warzybok (Medical Phys. and Acoust., Cluster of Excellence Hearing4All, Univ. of Oldenburg, Ammerländer Heerstr. 114-118, Oldenburg D-26129, Germany, thomas.brand@uni-oldenburg.de), Jan Rennies (Hearing, Speech and Audio Technol., Fraunhofer IDMT , Oldenburg, Germany), and Birger Kollmeier (Medical Phys. and Acoust., Cluster of Excellence Hearing4All, Univ. of Oldenburg, Oldenburg, Germany)

Many models of speech intelligibility in rooms are based on monaural measures. However, the effect of binaural unmasking improves speech intelligibility substantially. The binaural speech intelligibility model (BSIM) uses multi-frequency-band equalization-cancellation (EC), which models human binaural noise reduction, and the Speech-Intelligibility-Index (SII), which calculates the resulting speech intelligibility. The model analyzes the signal-to-noise ratios at the left and the right ear (modeling better-ear-listening) and the interaural cross correlation of target speech and binaural interferer(s). The effect of the hearing threshold is modeled by assuming two uncorrelated threshold simulation noises for both ears. BSIM describes the (binaural) aspects of useful and detrimental room reflections, reverb, and background noise. Especially the interaction of delay time and direction of speech reflections with binaural unmasking in different acoustical situations was modeled successfully. BSIM can use either the binaural room impulse responses of speech and interferers together with their frequency spectra or binaural recordings of speech and noise. A short-term version of BSIM can be applied to modulated maskers and predicts the consequence of dip listening. Aspects of informational masking are not taken into account yet. To model different degrees of informational masking. the SII threshold has to be re-calibrated.

Contributed Papers

3:20

4pAA7. Investigation of speech privacy in high-speed train cabins using a 1:10 scale model. Hansol Lim, Hyung Suk Jang, and Jin Yong Jeon (Architectural Eng., Hanyang Univ., 605-1, Sci. Technol. Bldg., Hang-dang dong, Seon-dong gu, Seoul KS013, South Korea, lim90128@gmail.com)

In this study, a 1:10 scale model was used to evaluate the acoustical parameters and speech transmission indices in high-speed train cabins when the interior design factors are changed to improve speech privacy. The 1:10 scale model materials were selected by considering real measured target factors, such as reverberation time (RT) and speech level (Lp,A,s). The characteristics of the background noise in a high-speed train depend on the train's speed; therefore, recordings of the background noise (LAeq) inside a train were considered in three situations: a stopped train, a train traveling at 100 km/h, and a train traveling at 300 km/h. The values of the STI were reproduced with the background noise levels at each speed using external array speakers with an equalizing filter in the scale model. The shapes and absorptions of chairs and interior surfaces were evaluated using scale modeling.

3:35

4pAA8. Laboratory experiments for speech intelligibility and speech privacy in passenger cars of high speed trains. Sung Min Oh, Joo Young Hong, and Jin Yong Jeon (Architectural Eng., Hanyang Univ., No. 605-1, Science&y Bldg., 222 Wangsimni-ro, Seongdong-gu, Seoul 133791, South Korea, pdpd5@naver.com)

This study explores the speech privacy criteria in passenger cars of high-speed trains. *In-situ* measurements were performed in running trains to analyze the acoustical characteristics of interior noises in train cabins, and laboratory experiments were conducted to determine the most appropriate single-number quantity for the assessment of speech privacy. In the listening tests, the participants were asked to rate (1) speech intelligibility, (2) speech privacy, and (3) annoyance with varying background noises and signal to noise ratio (SNR). From the results of the listening tests, the effects of background noise levels and SNR on the speech privacy and annoyance were examined and the optimum STI and background noise levels in the passenger car concerning both speech privacy and annoyance were derived.

3:50

4pAA9. Some effects of reflections and delayed sound arrivals on the perception of speech and corresponding measurements of the speech transmission index. Peter Mapp (Peter Mapp Assoc., Copford, Colchester CO6 1LG, United Kingdom, petermapp@petermapp.com)

Although the effects of reflections and later arriving sound repetitions (echoes) have been well researched and published over the past 60 years—ranging from Haas, Wallach, and more recently to Bradley & Sato and Toole, their effect on Speech Transmission Index measurements and assessments has only be cursorily studied. Over the past 20 years, the speech Transmission Index (STI) has become the most widely employed measure of potential speech intelligibility for both natural speech and more importantly of Public Address and emergency sound systems and Voice Alarms. There is a common perception that STI can fully account for echoes and late, discrete sound arrivals and reflections. The paper shows this not to be the case but that sound systems achieving high STI ratings can exhibit poor and unacceptable speech intelligibility due to the presence of late sound arrivals and echoes. The finding is based on the results of a series of listening tests and extensive sound system modeling, simulations and measurements. The results of the word score experiments were found to be highly dependent upon the nature of the test material and presentation.

4:05

4pAA10. Effects of room-acoustic exposure on localization and speech perception in cocktail-party listening situations. Renita Sudirga (Health and Rehabilitation Sci. Program, Western Univ., Elborn College, London, ON N6G 1H1, Canada, rsudirga@uwo.ca), Margaret F. Cheesman, and Ewan A. Macpherson (National Ctr. for Audiol., Western Univ., London, ON, Canada)

Given previous findings suggesting perceptual mechanisms counteracting the effects of reverberation in a number of listening tasks, we asked whether listening experience in a particular room can enhance localization and speech perception abilities in cocktail-party situations. Utilizing the CRM stimuli we measured listeners' abilities in: (1) identifying the location of a speech target given a (-22.5^o, 0^o, $+$22.5^o) talker configuration, (2) identifying the target color/number under co-located (0^o, 0^o, 0^o) and spatially-separated (Ő22.5^o, 0^o, $+$22.5^o) configurations. Stimuli were presented in three types of artificial reverberation. All reverberation types had the same relative times-of-arrival and levels of the reflections (T_{60} = 400 ms, C_{50} = 14 dB; wideband) and varied only in the lateral spread of the reflections. Reverberated stimuli were presented via a circular loudspeaker array situated in an anechoic chamber. Listening exposure was varied by mixing or fixing the reverberation type within a block of trials. For the location identification task, exposure benefit decreased with increasing Target-to-Masker Ratio (TMR). No exposure effect was observed in the speech perception task at Ő4 to 10 dB TMRs, except in the separated, narrowest reverberation condition. Results will be discussed in relation to the different nature of the tasks and findings from other studies.

4:20

4pAA11. On the use of a real-time convolution system to study perception of and response to self-generated speech and music in variable acoustical environments. Jennifer K. Whiting, Timothy W. Leishman (Dept. of Phys. and Astronomy, Brigham Young Univ., C110 ESC, Brigham Young University, Provo, UT 84606, lundjenny@comcast.net), and Eric J. Hunter (College of Commnication Arts & Sci., Michigan State Univ., East Lansing, MI)

A real-time convolution system has been developed to quickly manipulate the auditory room-acoustical experiences of human subjects. This system is used to study the perception of self-generated speech and music and the responses of talkers and musicians to varying conditions. Simulated and measured oral-binaural room impulse responses are used within the convolution system. Subjects in an anechoic environment experience room responses excited by their own voices or instruments via the convolution system. Direct sound travels directly to the ear, but the convolved room response is heard specialized headphones spaced away from the head. The convolution system, a method for calibrating room level to be consistent across room impulse responses, and data from preliminary testing for vocal effort in various room environments are discussed.

4:35

4pAA12. Use of k-means clustering analysis to select representative head related transfer functions for use in subjective studies. Matthew Neal and Michelle C. Vigeant (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, mtn5048@psu.edu)

A head related transfer function (HRTF) must be applied when creating auralizations; however, the HRTFs of individual subjects are not typically known in advance. Often, an overall 'average' HRTF is used instead. The purpose of this study was to develop a listening test to identify a 'matched' (best) and 'unmatched' (worst) HRTF for specific subjects, which could be applied to customize auralizations for individual participants. The method of k-means clustering was used to identify eight representative HRTFs from the CIPIC database. HRTFs from 45 subjects' left and right ears in four directions were clustered, which resulted in 56 cluster centers (possible representative HRTFs). A comparative analysis was conducted to determine an appropriate set of HRTFs. These HRTFs were then convolved with pink noise bursts at 00 elevation and various azimuths to sound like the bursts were rotating around a subject's head. A paired comparison test was used where listeners selected the 'most natural' sounding HRTF signal. 'Most natural' was described as coming from the correct directions and located outside the head. The results from the clustering analysis and listening test will be presented, along with a subjective study that incorporated the HRTF listening test. [Work was supported by NSF Grant 1302741.]

Session 4pAB

Animal Bioacoustics: Acoustics as a Tool for Population Structure III

Shannon Rankin, Cochair
Southwest Fisheries Science Ctr., 8901 La Jolla Shores Dr., La Jolla, CA 92037

Kathleen Stafford, Cochair
Appl. Phys. Lab., Univ. of Washington, 1013 NE 40th St., Seattle, WA 98105

Contributed Papers

1:15

4pAB1. Improving acoustic time-of-arrival location estimates by correcting for temperature drift in time base oscillators. Harold A. Cheyne, Peter M. Marchetto, Raymond C. Mack, Daniel P. Salisbury, and Janelle L. Morano (Lab of Ornithology, Cornell Univ., 95 Brown Rd., Rm. 201, Ithaca, NY 14850, haroldcheyne@gmail.com)

Using multiple acoustic sensors in an array for estimating sound source location relies on time synchrony among the devices. When independent time synchrony methods—such as GPS time stamps—are unavailable, the precision of the time base in individual sensors becomes one of the main sources of error in synchrony, and consequently increases the uncertainty of location estimates. Quartz crystal oscillators, on which many acoustic sensors base sampling rate timing, have a vibration frequency that varies with temperature f(T). Each oscillator exhibits a different frequency-temperature relationship, leading to sensor-dependent sample rate drift. Our Marine Autonomous Recording Units (MARUs) use such oscillators for their sample rate timing, and they experience variations in temperature of at least 20°C between preparation in air and deployment underwater, leading to sample rate drift over their deployments. By characterizing each MARU's oscillator f(T) function, and measuring the temperature of the MARU during the deployment, we developed a post-processing method of reducing the sample rate drift. When applied to acoustic data from an array of MARUs, this post-processing method resulted in a statistically significant decrease of the mean sample rate drift by a factor of two, and subsequent lower errors in acoustically derived location estimates.

1:30

4pAB2. Acoustic scene metrics for spatial planning. Kathleen J. Vigness-Raposa, Adam S. Frankel, Jennifer Giard, Kenneth T. Hunter, William T. Ellison (Marine Acoust., Inc., 809 Aquidneck Ave., Middletown, RI 02842, kathleen.vigness@marineacoustics.com)

Potential effects of anthropogenic underwater sounds on marine mammals are usually assessed on the basis of exposure to one sound source. Recently published research modeling underwater noise exposure and assessing its impact on marine life has extended the typical single source/single species absolute received level approach to defining exposure in a variety of ways including: relative levels of exposure, such as loudness, signal to noise ratio, and sensation level; metrics for evaluating chronic elevation in background noise; cumulative exposure to multiple and dissimilar sound sources, as well as the potential for animals to selectively avoid a particular source and other behavioral changes. New approaches to managing the overall acoustic scene that account for these issues requires a more holistic and multi-dimensional approach that addresses the relationships among the noise environment, animal hearing and behavior, and anthropogenic sound sources. We present a layered acoustic scene concept that considers each facet of the extended problem. Our exemplar is a seismic survey in the Gulf of Mexico with layers for ambient oceanographic and meteorological noise, shipping, and distant anthropogenic sources in which the exposure is filtered by the animal's hearing filter, sensation level, and nominal loudness of the signal.

1:45

4pAB3. Establishing baselines for cetaceans using passive acoustic monitoring off west Africa. Melinda Rekdahl, Salvatore Cerchio, and Howard Rosenbaum (WCS, 2300 Southern Blvd., The Bronx, New York, NY 10460, mlrekdahl@wcs.org)

Knowledge of cetacean presence in west African waters is sparse due to the remote and logistically challenging nature of working in these waters. Exploration and Production (E&P) activities are increasing in this region; therefore, collecting baseline information on species distribution is important. Previous research is limited although a number of species listed as vulnerable or data deficient by the IUCN red list have been documented. In 2012/2013, we deployed an array of eight Marine Autonomous Recording Units (MARUs) in a series of three deployments, off Northern Angola, targeting Mysticetes (2 kHz SR, continuous) during winter/spring and Odontocetes (32 kHz SR, 20% duty cycled) during summer/autumn. Preliminary results are presented on the temporal and spatial distribution of species identified from automated and manual detection methods. Humpback whales were frequently detected from August through December, with peaks during September/October. During the deployment period, sperm whales and Balaenopterid and Odontocete calls were also detected and possible species will be discussed. Species detections will be used to identify temporal hotspots for cetacean presence and any potential overlap with E&P activities. We recommend that future research efforts include visual and acoustic vessel surveys to increase the utility of passive acoustics for monitoring these populations.

2:00

4pAB4. Behavioral response of select reef fish and sea turtles to mid-frequency sonar. Stephanie L. Watwood, Joseph D. Iafrate (NUWC Newport, 1176 Howell St., Newport, RI 02841, stephanie.watwood@navy.mil), Eric A. Reyier (Kennedy Space Ctr. Ecological Program, Kennedy Space Ctr., FL), and William E. Redfoot (Marine Turtle Res. Group, Univ. of Central Florida, Orlando, FL)

There is growing concern over the potential effects of high-intensity sonar on wild marine species populations and commercial fisheries. Acoustic telemetry was employed to measure movements of free-ranging reef fish and sea turtles in Port Canaveral, Florida, in response to routine submarine sonar testing. Twenty-five sheepshead (*Archosargus probatocephalus*), 28 gray snapper (*Lutjanus griseus*), and 29 green sea turtles (*Chelonia mydas*) were tagged, with movements monitored for a period of up to four months using an array of passive acoustic receivers. Baseline residency was examined for fish and sea turtles before, during, and after the test event. No mortality of tagged fish or sea turtles was evident from the sonar test event. There was a significant increase in daily residency index for both sheepshead and gray snapper at the testing wharf subsequent to the event. No broad-scale movement from the study site was observed during or immediately after the test. One month after the sonar test, 56% of sheepshead, 71% of gray snappers, and 24% of green sea turtles were still detected on receivers located at the sonar testing wharf.

2:15

4pAB5. Quantifying the ocean soundscape at a very busy southern California location. John E. Joseph and Tetyana Margolina (Oceanogr., Naval Postgrad. School, 833 Dyer Rd, Monterey, CA 93943, jejoseph@nps.edu)

The underwater noise environment in the Southern California Bight is highly variable due to the presence of both episodic and persistent contributors to the soundscape. Short-term events have potential for inducing abrupt behavioral responses in marine life while long-term exposure may have chronic influences or cause more subtle responses. Here we identify and quantify various sources of sound over a wide frequency band using a passive acoustic receiver deployed at 30-mi Bank from December 2012 through March 2013. The site is in the eastern portion of the Navy's training range complex and is in close proximity to very active shipping routes. The region has diverse marine habitats and is known for frequent seismic activity. Acoustic data were scanned for anthropogenic, biologic and other natural noise sources up to 100 kHz. In addition, ancillary databases and data sets were used to verify, supplement and interpret results. Acoustic propagation models were used to explain ship-induced noise patterns. Results indicate that long-term trends in soundscapes over regional-scale areas can be accurately estimated using a combination of tuned acoustic modeling and recurrent *in-situ* data for validation. [Project funded by US Navy.]

2:30

4pAB6. Machine learning an audio taxonomy: Quantifying biodiversity and habitat recovery through rainforest audio recordings. Tim Treuer (Ecology and Evolutionary Biology, Princeton Univ., Princeton, NJ), Jaan Altosaar, Andrew Hartnett (Phys., Princeton Univ., 88 College Rd. West, Princeton, NJ 08544, altosaar@princeton.edu), Colin Twomey, Andy Dobson, David Wilcove, and Iain Couzin (Ecology and Evolutionary Biology, Princeton Univ., Princeton, NJ)

We present a set of tools for semi-supervised classification of ecosystem health in Meso-American tropical dry forest, one of the most highly endangered habitats on Earth. Audio recordings were collected from 15-year-old, 30-year-old and old growth tropical dry forest plots in the Guanacaste Conservation Area, Costa Rica, on both nutrient rich and nutrient poor soils. The goals of this project were to classify the overall health of the regenerating forests using markers of biodiversity. Semi-supervised machine learning and digital signal processing techniques were explored and tested for their ability to detect species and events in the audio recordings. Furthermore, multi-recorder setups within the same vicinity were able to improve detection rates and accuracy by enabling localization of audio events. Variations in species' and rainforest ambient noise detection rates over time were hypothesized to correlate to biodiversity and hence the health of the rainforest. By comparing levels of biodiversity measured in this manner between old growth and young dry forest plots, we hope to determine the effectiveness of reforestation techniques and identify key environmental factors shaping the recovery of forest ecosystems.

2:45–3:00 Break

3:00

4pAB7. Sound-based automatic neotropical sciaenid fishes identification: *Cynoscion jamaicensis*. Sebastian Ruiz-Blais (Res. Ctr. of Information and Commun. Technologies, Universidad de Costa Rica, Guadalupe, Goicoechea, San José 1385-2100, Costa Rica, ruizble@yahoo.com), Arturo Camacho (School of Comput. Sci. and Informatics, Universidad de Costa Rica, San José, Costa Rica), and Mario R. Rivera-Chavarria (Res. Ctr. of Information and Commun. Technologies, Universidad de Costa Rica, San José, Costa Rica)

Automatic software for sciaenid sound emissions identification are scarce. We present a method to automatically identify sound emissions produced by the sciaenid *Cynoscion jamaicensis*. The emissions of *C. jamaicensis* typically have a 24 Hz pulse repetition rate and a quasi-harmonic pattern in their spectra with a pitched quality in its sound. The proposed method is an adaptation of a previous method proposed to detect sounds of *Cynoscion squamipinnis* in recordings. It features long-term partial loudness, pulse repetition rate, pitch strength, and timbre statistics. The satisfactory results of 0.9 in the F-measure show that the method generalizes well over species, considering the different characteristics of *C. jamaicensis* and *C. squamipinnis*. Future research is required to test the method with other species recordings, in order to further evaluate its robustness.

3:15

4pAB8. Examining the impact of the ocean environment on cetacean classification using theocean acoustics and seismic exploration synthesis (OASES) propagation model. Carolyn M. Binder and Paul C. Hines (Defence R&D Canada , P.O. Box 1012, Dartmouth, NS B2Y 3Z7, Canada, carolyn.binder@drdc-rddc.gc.ca)

Passive acoustic monitoring (PAM) is now in wide use to study cetaceans in their natural habitats. Since cetaceans can be found in all ocean basins, their habitats cover diverse underwater environments. Properties of the ocean environment such as the sound speed profile, bathymetry, and sediment properties can be markedly different between these diverse environments. This leads to differences in how a cetacean vocalization is distorted by propagation effects and may impact the accuracy of PAM systems. To develop an automatic PAM system capable of operating effectively under numerous environmental conditions one must understand how propagation conditions affect these systems. Previous effort using a relatively limited data set has shown that a prototype aural classifier developed at Defence R&D Canada can be used to reduce false alarm rates and successfully discriminate cetacean vocalizations from several species. The aural classifier achieves accurate results by using perceptual signal features that model the features employed by the human auditory system. The current work uses the OASES pulse propagation model to examine the robustness of the classifier under various environmental conditions; preliminary results will be presented from cetacean vocalizations that were transmitted over several ranges through environments modeled using conditions measured during experimental trials.

3:30

4pAB9. Acoustic detection, localization, and tracking of vocalizing humpback whales on the U.S. Navy's Pacific Missile Range Facility. Tyler A. Helble (SSC-PAC, 2622 Lincoln Ave., San Diego, CA 92104, tyler.helble@gmail.com)

A subset of the 41 deep water broadband hydrophones on the U.S. Navy's Pacific Missile Range Facility (PMRF) to the northwest of Kauai, Hawaii was used to acoustically detect, localize, and track vocalizing humpback whales as they transited through this offshore range. The focus study area covers 960 square kilometers of water (water depths greater than 300 m and more than 20 km offshore). Because multiple animals vocalize simultaneously, novel techniques were developed for performing call association in order to localize and track individual animals. Several dozen whale track lines can be estimated over varying seasons and years from the hundreds of thousands of recorded vocalizations. An acoustic model was used to estimate the transmission loss between the animal and PMRF hydrophones so that source levels could be accurately estimated. Evidence suggests a Lombard effect: the average source level of humpback vocalizations changes with changes in background noise level. Additionally, song bout duration, cue (call) rates, swim speeds, and movement patterns of singing humpback whales can be readily extracted from the track estimates. [This work was supported by Commander U.S. Pacific Fleet, the Office of Naval Research, and Living Marine Resources.]

3:45

4pAB10. Determining the detection function of passive acoustic data loggers for porpoises using a large hydrophone array. Jens C. Koblitz, Katharina Brundiers, Mario Kost (German Oceanogr. Museum, Katharinenberg 14-20, Stralsund 18439, Germany, Jens.Koblitz@meeresmuseum.de), Louise Burt, Len Thomas (Ctr. for Res. into Ecological and Environ. Modelling, Univ. of St. Andrews, St. Andrews, United Kingdom), Jamie MacAulay (Sea Mammal Res. Unit, Univ. of St. Andrews, St. Andrews, United Kingdom), Cinthia T. Ljungqvist (Kolmarden Wildlife Park, Kolmarden, Sweden), Lonnie Mikkelsen (Dept. of BioSci., Aarhus Univ., Roskilde, Denmark), Peter Stilz (Freelance Biologist, Hechingen, Germany), and Harald Benke (German Oceanogr. Museum, Stralsund, Germany)

Click loggers such as C-PODs are an important tool to monitor the spatial distribution and seasonal occurrence of small odontocetes. To determine absolute density, information on the detection function, the detection probability as a function of distance, and derived from this, the effective detection radius (EDR), is needed. In this study a 15 channel hydrophone array, deployed next to 12 C-PODs, was used to localize porpoises and determine their geo-referenced swim paths using the ship's GPS and motion sensors. The detection function of C-PODs was then computed using the distance between the animals and each C-POD. In addition to this, the acoustic detection function of C-PODs has been measured by playing back porpoise-like clicks using an omni-directional transducer. The EDR for these porpoise-like clicks with a source level of 168 dB re 1 μPa pp varied from 41 to 243 m. This variation seemed to be related to the sensitivity of the devices; however, season and water depth also seemed to have an influence on detectability.

4:00

4pAB11. Variations of soundscape in a shallow water marine environment for the Chinese white dolphin. Shane Guan (Dept. of Mech. Eng., The Catholic Univ. of America, 1315 East-West Hwy., SSMC-3, Ste. 13700, Silver Spring, MD 20902, shane.guan@noaa.gov), Tzu-Hao Lin, Lien-Siang Chou (Inst. of Ecology and Evolutionary Biology, National Taiwan Univ., Taipei, Taiwan), and Joseph F. Vignola (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, MD)

For acoustically oriented animals, sound field can either provide or mask critical information for their well-being and survival. In addition, understanding the variations of the soundscape in the Chinese white dolphin habitat is important to monitoring the relationship between human activities, calling fish, and dolphins, thus assist in coastal conservation and management. Here, we examined the soundscape of a critically endangered Chinese white dolphin population in two shallow water areas next to western coast of Taiwan. Two recording stations were established at Yunlin, which is close to an industrial harbor, and Waisanding, which is nearby a fishing village, in summer 2012. Site specific analyses were performed on variations of the temporal and spectral acoustic characteristics for both locations. The results show different soundscapes for the two sites from different recurring human activities. At Yunlin, high acoustic energy was usually dominated by cargo ships producing noise below 1 kHz. At Waisanding, much higher frequency noise, up to 16 kHz produced by passing fishing boats were detected. In addition, a diurnal cycle of the acoustic field between 1200 and 2600 Hz was observed. It is established that this sound was produced by fish chorus that were observed in both locations.

4:15

4pAB12. Anthropogenic noise has a knock-on effect on the behavior of a territorial species. Kirsty E. McLaughlin and Hansjoerg P. Kunc (School Biological Sci., Queens Univ. Belfast, 97 lisburn Rd., MBC, Belfast bt9 7gt, United Kingdom, kmclaughlin23@qub.ac.uk)

Noise pollution has been shown to induce overt behavioral changes such as avoidance of a noise source and changes to communication behavior. Few studies however have focused on the more subtle behaviors within an individual's repertoire such as foraging and territoriality. Many species are territorial making it unlikely they will leave a noisy area. The impact of noise on essential behaviors of such species must be examined. It has been suggested that a noise induced increase in sheltering behavior will decrease time available for other activities. To test for this potential knock-on effect, we exposed a territorial fish to noise of differing sound pressure levels (SPL). We found that exposure to noise increased sheltering behavior and decreased foraging activity. However, we found that these behavioral responses did not increase with SPL. Furthermore we demonstrate, for the first time experimentally, that noise has a negative knock-on effect on behavior as a noise induced increase in sheltering caused a decrease in foraging activity. This novel finding highlights the importance of examining less overt behavioral changes caused by noise, especially in those species unlikely to avoid a noisy area, and suggests the impacts of noise on animals may be greater than previously predicted.

4:30

4pAB13. Female North Atlantic right whales produce gunshot sounds. Edmund R. Gerstein (Psych., Florida Atlantic Univ., 777 Glades Rd., Boca Raton, FL 33486, gerstein2@aol.com), Vailis Trygonis (FAU / Harbor Branch Oceanogr. Inst. , Lesvos island, Greece), Steve McCulloch (FAU / Harbor Branch Oceanogr. Inst. , Fort Pierce, FL), Jim Moir (Marine Resources Council , Stuart, FL), and Scott Kraus (Edgerton Res. Lab., New England Aquarium, Bostopn, MA)

North Atlantic right whales (*Eubalaena glacialis*) produce loud, broadband, short duration sounds referred to as gunshots. The sounds have been hypothesized to function in a reproductive context, as sexual advertisement signals produced by solitary adult males to attract females and/or agonistic displays among males in surface active groups. This study provides evidence that gunshot sounds are also produced by adult females and examines the acoustics and behavioral contexts associated with these calls. Results from boat-based observational surveys investigating the early vocal ontogeny and behavior of right whales in the critical southeast calving habitat are presented for a subset of mothers who produced gunshots while in close proximity to their calves. Of 26 different isolated mother-calf pairs, gunshots were recorded from females of varied ages and maternal experience. The signals were recorded when calves separated from their mothers during curious approaches toward objects on the surface. While the spectral and temporal characteristics of female gunshots resemble those attributed to adult males, these calls were orders of magnitude quieter (Ő30 dB). Relatively quiet gunshots posed minimal risk of injury to nearby calves. The social and behavioral context suggests gunshots were associated with maternal communication and may also be indicators of stress and agitation.

4:45

4pAB14. Classifying humpback whale individuals from their nocturnal feeding-related calls. Wei Huang, Fan Wu (Elec. and Comput. Eng., Northeastern Univ., 360 Huntington Ave., 302 Stearns, Boston, MA 02115, weihece@gmail.com), Nicholas C. Makris (Mech. Eng., Massachusetts Inst. of Technol., Cambridge, MA), and Purnima R. Makris (Elec. and Comput. Eng., Northeastern Univ., Boston, MA)

A large number of humpback whale vocalizations, comprising of both songs and non-song calls, were passively recorded on a high-resolution towed horizontal receiver array during a field experiment in the Gulf of Maine near Georges Bank in the immediate vicinity of the Atlantic herring spawning ground from September to October 2006. The non-song calls were highly nocturnal and dominated by trains of "meows," which are downsweep chirps lasting roughly 1.4 s in the 300 to 600 Hz frequency range, related to night-time foraging activity. Statistical temporal-spectral analysis of the downsweep chirps from a localized whale group indicate that these "meows"can be classified into six or seven distinct types that occur repeatedly over the nighttime observation interval. These meows may be characteristic of different humpback individuals, similar to human vocalizations. Since the "meows" are feeding-related calls for night-time communication or prey echolocation, they may originate from both adults and juveniles of any gender; whereas songs are uttered primarily by adult males. The meows may then provide an approach for passive detection, localization and classification of humpback whale individuals regardless of sex and maturity, and be especially useful for night-time and/or long range monitoring and enumeration of this species.

Session 4pBAa

Biomedical Acoustics: Biomedical Applications of Low Intensity Ultrasound II

Thomas L. Szabo, Chair
Biomedical Dept., Boston Univ., 44 Cummington Mall, Boston, MA 02215

Contributed Papers

1:00

4pBAa1. Investigation of effects of ultrasound on dermal wound healing in diabetic mice. Denise C. Hocking (Pharmacology and Physiol., Univ. of Rochester, 601 Elmwood Ave., Box 711, Rochester, NY 14642, denise_hocking@urmc.rochester.edu), Carol H. Raeman, and Diane Dalecki (Biomedical Eng., Univ. of Rochester, Rochester, NY)

Chronic wounds, including diabetic, leg, and pressure ulcers, impose a significant health care burden worldwide. Currently, chronic wound therapy is primarily supportive. Ultrasound therapy is used clinically to promote bone healing and some evidence indicates that ultrasound can enhance soft tissue repair. Here, we investigated effects of ultrasound on dermal wound healing in a murine model of chronic wounds. An ultrasound exposure system and protocol were developed to provide daily ultrasound exposures to full-thickness, excisional wounds in genetically diabetic mice. Punch biopsy wounds were made on the dorsal skin and covered with acoustically transparent dressing. Mice were exposed to 1-MHz pulsed ultrasound (2 ms pulse, 100 Hz PRF, 0–0.4 MPa) for a duration of 8 min per day. Mice were exposed on 10 days over a 2-week period. No significant differences in the rate of re-epithelialization were observed in response to ultrasound exposure compared to sham-exposed controls. However, two weeks after injury, a statistically significant increase in granulation tissue thickness at the wound center was observed in mice exposed to 0.4 MPa (389 +/ 85 μm) compared to sham exposures (105 +/Ő 50 μm). Additionally, histological sections showed increased collagen deposition in wounds exposed to 0.4 MPa compared to shams.

1:15

4pBAa2. Evaluation of sub-micron, ultrasound-responsive particles as a drug delivery strategy. Rachel Myers, Susan Graham, James Kwan, Apurva Shah, Steven Mo, and Robert Carlisle (Inst. of Biomedical Eng., Univ. of Oxford, Dept. of Eng. Sci., ORCRB, Headington, Oxford OX3 7DQ, United Kingdom, rachel.myers@eng.ox.ac.uk)

Substantial portions of tumors are largely inaccessible to drugs due to their irregular vasculature and high intratumoral pressure. The enhanced permeability and retention effect causes drug carriers within the size range of 100–800 nm to passively accumulate within tumors; however, they remain localized close to the vasculature. Failure to penetrate into and throughout the tumor ultimately limits treatment efficacy. Ultrasound-induced cavitation events have been cited as a method of stimulating greater drug penetration. At present, this targeting strategy is limited by the difference in size between the nano-scale drug carriers used and the cavitation nuclei available, i.e., the micron-scale contrast agent SonoVue. *In vivo* this results in spatial separation of the two agents, limiting the capacity for one to impact upon the other. Our group has successfully formulated two different monodisperse suspensions of nanoparticles that are of a size that will permit better co-localization of cavitation nuclei and therapeutics. A mixture of these nanoparticles and a model drug carrier were passed through a tissue mimicking phantom to provide an *in vitro* simulation of flow through a tumor. The impact of ultrasound on the penetration of drug carrier from the flow channel was compared between both of our ultrasound-responsive particles and SonoVue.

1:30

4pBAa3. Temperature effects on the dynamics of contrast enhancing microbubbles. Faik C. Meral (Radiology, Brigham and Women's Hospital, 221 Longwood Ave., EBRC 521, Boston, MA 02115, fcmeral@bwh.harvard.edu)

Micron-sized, gas encapsulated bubbles are used as ultrasound contrast enhancing agents to improve diagnostic image quality. These microbubbles, which are vascular agents, undergo linear and non-linear oscillations when excited. It is this non-linear response of microbubbles, that helps to distinguish between signals from the tissue -mostly linear-, and signals from the bubbles,nonlinear, which represents vasculature. This opens up to numerous clinical applications such as echocardiography, focal lesion identification, perfusion imaging, etc. Characterization studies of microbubbles gained importance as the possible clinical applications increase. One aspect that these studies focused on is the temperature dependence of the microbubble dynamics. However, these studies were mostly comparing bubble dynamics at room temperature to their dynamics at the physiological temperatures. This study is focused on the changes in the bubble characteristics as a function of temperature. More specifically microbubble attenuation and scattering is measured as a function of temperature and time. Additionally, estimating the temperature changes from the changes in the bubble dynamics is considered as an inverse problem.

1:45

4pBAa4. Response to ultrasound of two types of lipid-coated microbubbles observed with a high-speed optical camera. Tom van Rooij, Ying Luan, Guillaume Renaud, Antonius F. W. van der Steen, Nico de Jong, and Klazina Kooiman (Dept. of Biomedical Eng., Erasmus MC, Postbus 2040, Rotterdam 3000 CA, Netherlands, t.vanrooij@erasmusmc.nl)

Microbubbles (MBs) can be coated with different lipids, but exact influences on acoustical responses remain unclear. The distribution of lipids in the coating of homemade MBs is heterogeneous for DSPC and homogeneous for DPPC-based MBs, as observed with 4Pi confocal microscopy. In this study, we investigated whether DSPC and DPPC MBs show a different vibrational response to ultrasound. MBs composed of main lipid DSPC or DPPC (2 C-atoms less) with a C_4F_{10} gas core, were made by sonication. Microbubble spectroscopy was performed by exciting single MBs with 10-cycle sine wave bursts having a frequency from 1 to 4 MHz and a peak negative pressure of 10, 20, and 50 kPa. The vibrational response to ultrasound was recorded with the Brandaris 128 high-speed camera at 15 Mfps. Larger acoustically induced deflation was observed for DPPC MBs. For a given resting diameter, the resonance frequency was higher for DSPC, resulting in higher shell elasticity of 0.26 N/m as compared to 0.06 N/m for DPPC MBs. Shell viscosity was similar (~10^{-8} kg/s) for both MB types. Non-linear behavior was characterized by the response at the subharmonic and second harmonic frequencies. More DPPC (71%) than DSPC MBs (27%) showed subharmonic response, while the behavior at the second harmonic frequency was comparable. The different acoustic responses of DSPC and DPPC MBs are likely due to the choice of the main lipid and the corresponding spatial distribution in the MB coating.

2:00

4pBAa5. Quantitative acoustic microscopy at 250 MHz for unstained *ex vivo* assessment of retinal layers. Daniel Rohrbach (Lizzi Ctr. for Biomedical Eng., Riverside Res. Inst., 156 William St., 9th Fl., New York City, NY 11215, drohrbach@RiversideResearch.org), Harriet O. Lloyd, Ronald H. Silverman (Dept. of Ophthalmology, Columbia Univ. Medical Ctr., New York City, NY), and Jonathan Mamou (Lizzi Ctr. for Biomedical Eng., Riverside Res. Inst., New York City, NY)

Few quantitative acoustic microscopy (QAM) investigations have been conducted on the vertebrate retina. However, quantitative assessment of acoustically-related material properties would provide valuable information for investigating several diseases. We imaged 12-μm sections of deparaffinized eyes of rdh4 knockout mice (N = 3) using a custom-built acoustic microscope with an F-1.16, 250-MHz transducer (Fraunhofer IBMT) with a 160-MHz bandwidth and 7-μm lateral beamwidth. 2D QAM maps of ultrasound attenuation (UA) and speed of sound (SOS) were generated from reflected signals. Scanned samples then were stained using hematoxylin and eosin and imaged by light microscopy for comparison with QAM maps. Spatial resolution and contrast of QAM maps of SOS and UA were sufficient to resolve anatomic layers within the 214 μm thick retina; anatomic features in QAM maps corresponded to those seen by light microscopy. UA was significantly higher in the outer plexiform layer (420$\pm$70 dB/mm) compared to the inner nuclear layer (343$\pm$22 dB/mm). SOS values ranged between 1696$\pm$56 m/s for the inner nuclear layer and 1583$\pm$42 m/s for the inner plexiform layer. To the authors' knowledge, this study is the first to assess the UA, and SOS of retina layers of vertebrate animals at high frequencies. [NIH Grant R21EB016117 and Core Grant P30EY019007.]

2:15

4pBAa6. Acoustic levitation of gels: A proof-of-concept for thromboelastography. Nate Gruver and R. Glynn Holt (Dept. of Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, Nate_Gruver@buacademy.org)

Current thromboelastography in the clinic requires contact between the measurement apparatus and the blood being studied. An alternative technique employs levitation of a small droplet to limit contact with the blood sample to air alone. As has been demonstrated for Newtonian liquid drops, the measurement of static spatial location and sample deformation can be used to infer sample surface tension. In the current study, ultrasonic acoustic levitation was used to levitate viscoelastic samples. Gelatin was used as a stand-in for blood to establish the validity of the ultrasonic levitation technique on viscoelastic materials. Liquid data was first taken to benchmark the apparatus, then deformation/location studies were performed on set and setting gelatin gels. Relationships between gelling time, gel concentration, and gel firmness were demonstrated. The elastic modulus of gels was inferred from the data using an idealized model.

2:30

4pBAa7. Numerical simulations of ultrasound-lung interaction. Brandon Patterson (Mech. Eng., Univ. of Michigan, 1231 Beal Ave., Ann Arbor, MI 48109, awesome@umich.edu), Douglas L. Miller (Radiology, Univ. of Michigan, Ann Arbor, MI), David R. Dowling, and Eric Johnsen (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Lung hemorrhage (LH) remains the only bioeffect of non-contrast, diagnostic ultrasound (DUS) proven to occur in mammals. While DUS for lung imaging is routine in critical care situations, a fundamental understanding of DUS-induced LH remains lacking. The objective of this study is to numerically simulate DUS-lung interaction to identify potential damage mechanisms, with an emphasis on shear. Experimentally relevant ultrasound waveforms of different frequencies and amplitudes propagate in tissue (modeled as water) and interact with the lung (modeled as air). Different length scales ranging from single capillaries to lung surface sizes are investigated. For the simulations, a high-order accurate discontinuity-capturing scheme solves the two-dimensional, compressible Navier-Stokes equations to obtain velocities, pressures, stresses and interface displacements in the entire domain. In agreement with theoretical acoustic approximations, small interface displacements are observed. At the lung surface, shear stresses indicative of high strains rates develop and are shown to increase nonlinearly with decreasing ratio of interface curvature to ultrasonic wavelength.

THURSDAY AFTERNOON, 8 MAY 2014 BALLROOM E, 3:00 P.M. TO 5:30 P.M.

Session 4pBAb

Biomedical Acoustics: Modeling and Characterization of Biomedical Systems

Diane Dalecki, Chair

Biomedical Eng., Univ. of Rochester, 310 Goergen Hall, P.O. Box 270168, Rochester, NY 14627

Contributed Papers

3:00

4pBAb1. Green's function-based simulations of shear waves generated by acoustic radiation force in elastic and viscoelastic soft tissue models. Yiqun Yang (Dept. of Elec. and Comput. Eng., Michigan State Univ., East Lansing, MI), 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., 428 S. Shaw, 2120 Eng. Bldg., East Lansing, MI 48824, mcgough@egr.msu.edu)

The Green's function approach describes propagating shear waves generated by an acoustic radiation force in elastic and viscoelastic soft tissue. Calculations with the Green's function approach are evaluated in elastic and viscoelastic soft tissue models for a line source and for a simulated focused beam. The results for the line source input are evaluated at 200 time samples in a 101 by 101 point grid that is perpendicular to the line source. For a shear wave speed of 1.4832 m/s and a compressional wave speed of 1500 m/s, shear wave simulations for a line source input in elastic and viscoelastic soft tissue models are completed in 431 and 2487 s with MATLAB scripts, respectively, where the shear viscosity is 0.1 Pa.s in the viscoelastic model. Simulations are evaluated at a single point for an acoustic radiation force generated by a 128 element linear array operating at 4.09 MHz, and these simulations require 228 s and 1327 s for elastic and viscoelastic soft tissue models, respectively. The results show that these are effective models for simulating shear wave propagation in soft tissue, and plans to accelerate these simulations will also be discussed. [Supported in part by NIH Grants R01 EB012079 and R01 DK092255.]

3:15

4pBAb2. Improved simulations of diagnostic ultrasound with the fast nearfield method and time-space. Pedro C. Nariyoshi and Robert J. McGough (Dept. of Elec. and Comput. Eng., Michigan State Univ., 428 S. Shaw, 2120 Eng. Bldg., East Lansing, MI 48824, mcgough@egr.msu.edu)

Diagnostic ultrasound simulations are presently under development for FOCUS (http://www.egr.msu.edu/~fultras-web). To reduce the computation time without increasing the numerical error, each signal in FOCUS is calculated once, stored, and then the effects of different time delays are calculated with cubic spline interpolation. This is much more efficient than calculating the same transient signal at a scatterer repeatedly for different values of the time delay. Initially, the interpolation results were obtained from uniformly sampled signals, and now the signal start and end times are also considered. This step reduces the error in the pulse-echo calculation without significantly increasing the computation time. Simulated B-mode images were evaluated in a cyst phantom with 100 000 scatterers using this approach. Images with 50 A-lines are simulated for a linear array with 192 elements, where the translating subaperture contains 64 elements. The resulting simulated images are compared to images obtained with the same configuration in Field II (http://field-ii.dk/). An error of approximately 1% is achieved in FOCUS with a sampling frequency of 30 MHz, where Field II requires a sampling frequency of 180 MHz to reach the same error. FOCUS also reduces the simulation time by a factor of six. [Supported in part by NIH Grant R01 EB012079.]

3:30

4pBAb3. Simulations of ultrasound propagation in a spinal structure. Shan Qiao, Constantin-C Coussios, and Robin O. Cleveland (Dept. of Eng. Sci., University of Oxford, Biomedical Ultrason., Biotherapy & Biopharmaceuticals Lab. (BUBBL) Inst. of Biomedical Eng., Old Rd. Campus Res. Bldg., Headington, OXFORD, Oxford OX3 7DQ, United Kingdom, shan.qiao@eng.ox.ac.uk)

Lower back pain is one of the most common health problems in developed countries, the main cause of which is the structure change of the intervertebral disks due to the degeneration. High intensity focused ultrasound (HIFU) can be used to remove the tissue of the degenerate discs through acoustic cavitation, after which injection of a replacement material can restore normal physiological function. The acoustic pressure distribution in and around the disc is important for both efficiency and safety. Ultrasound propagation from two 0.5 MHz focused transducers (placed confocally and oriented at 90 degrees) were simulated using a three-dimensional finite element model (PZFlex, Wiedlinger Associates) for both a homogeneous medium and a bovine spine. The size of computation domain was 64 mm*95 mm*95 mm, with a mesh size of 15 elements per wavelength of the fundamental waveform. Measurements of the pressure field from the two transducers in water were also performed. The simulations in a homogeneous medium agreed with the experimental results, in which a sharp ultrasound focus was observed. However, for the spine, the interference of the vertebral bodies lead to absorption in the bone and a smearing of the focus. [Work supported by EPSRC.]

3:45

4pBAb4. Can quantitative synthetic aperture vascular elastography predict the stress distribution within the fibrous cap non-invasively. Steven J. Huntzicker and Marvin M. Doyley (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, huntzick@ece.rochester.edu)

An imaging system that can detect and predict the propensity of an atherosclerotic plaque to rupture would reduce stroke. Radial and circumferential strain elastograms can reveal vulnerable regions within the fibrous cap. Circumferential stress imaging could predict the propensity of rupture. However, circumferential stress imaging demands either accurate knowledge of the geometric location of the fibrous cap or high quality strain information. We corroborated this hypothesis by performing studies on simulated vessel phantoms. More precisely, we computed stress elastograms with (1) precise knowledge of the fibrous cap, (2) no knowledge of the fibrous cap, (3) imprecise knowledge of the fibrous cap. We computed stress elastograms with accuracy of 8%, 15%, and 23% from high precision axial and lateral strain elastograms (i.e., 25 dB SNR) when precise, imprecise, and no geometric information was included the stress recovery method. The stress recovery method produced erroneous elastograms at lower noise level (i.e., 15 dB SNR), when no geometric information was included. Similarly, it produced elastograms with accuracy of 13% and 30% when precise and imprecise geometric information was included. The stress imaging method described in this paper performs well enough to warrant further studies with phantoms and *ex-vivo* samples.

4:00

4pBAb5. Super wideband quantitative ultrasound imaging for trabecular bone with novel wideband single crystal transducer and frequency sweep measurement. Liangjun Lin, Eesha Ambike (Biomedical Eng., Stony Brook Univ., Rm. 212 BioEng. Bldg., 100 Nicolls Rd., Stony Brook, NY 11794-3371, john85726@gmail.com), Raffi Sahul (TRS, Inc., State College, PA), and Yi-Xian Qin (Biomedical Eng., Stony Brook Univ., Stony Brook, NY)

Current quantitative ultrasound (QUS) imaging technology for bone provides a unique method for evaluating both bone strength and density. The broadband ultrasound attenuation (BUA) has been widely accepted as a strong indicator for bone health status. Researchers have reported BUA data between 0.3 and 0.7 MHz have strong correlation with the bone density. Recently, a novel spiral-wrapped wideband ultrasound transducer fabricated from piezoelectric PMN-PT single crystal is developed by TRS. This novel transducer combines the piezoelectric single crystal material and use of wide-band resonance transducer to provide a bandwidth superior to commercial devices with the capacity for a high sensitivity. To evaluate its application in bone imaging, a trabecular bone plate (6.5 mm thick) was prepared. The TRS transducer emits customized chirp pulses through the bone plate. The bandwidth of the ultrasound pulses is 0.2 MHz, ranging from 0.2 to 3 MHz. Based on the attenuation of the received pulses, the frequency spectrum is created to analyze the attenuation characteristics of the ultrasound attenuation across the super wide bandwidth. This new transducer technology provides more information across a wider bandwidth than the conventional ultrasound transducer and can therefore give rise to new QUS modality to evaluate bone health status.

4:15

4pBAb6. Spectrum analysis of photoacoustic signals for characterizing lymph nodes. Parag V. Chitnis, Jonathan Mamou, and Ernest J. Feleppa (F. L. Lizzi Ctr. for Biomedical Eng., Riverside Res., 156 William St., 9th Fl., New York, NY 10038, pchitnis@riversideresearch.org)

Quantitative-ultrasound (QUS) estimates obtained from spectrum analysis of pulse-echo data are sensitive to tissue microstructure. We investigated the feasibility of obtaining quantitative photoacoustic (QPA) estimates for simultaneously providing sensitivity to microstructure and optical specificity, which could more robustly differentiate among tissue constituents. Experiments were conducted using four, gel-based phantoms ($1 \times 1 \times 2$ cm) containing black polyethylene spheres (1E5 particles/ml) that had nominal mean diameters of 23.5, 29.5, 42, or 58 μm. A pulsed, 532-nm laser excited the photoacoustic (PA) response. A 33-MHz transducer was raster scanned over the phantoms to acquire 3D PA data. PA signals were processed using rectangular-cuboidal regions-of-interests to yield three quantitative QPA estimates associated with tissue microstructure: spectral slope (SS), spectral intercept (SI), and effective-absorber size (EAS). SS and SI were computed using a linear-regression approximation to the normalized spectrum. EAS was computed by fitting the normalized spectrum to the multi-sphere analytical solution. The SS decreased and the SI increased with an increase in particle size. While EAS also was correlated with particle size, particle aggregation resulted in EAS estimates that were greater than the nominal particle size. Results indicated that QPA estimates potentially can be used for tissue classification. [Work supported by NIH grant EB015856.]

4:30

4pBAb7. Parametric assessment of acoustic output from laser-irradiated nanoparticle volumes. Michael D. Gray (School of Mech. Eng., Georgia Inst. of Technol., 771 Ferst Dr. NW, Atlanta, GA 30332-0405, michael.gray@me.gatech.edu), Aritra Sengupta, and Mark R. Prausnitz (School of Chemical and Biomolecular Eng., Georgia Inst. of Technol., Atlanta, GA)

A photoacoustic technique is being investigated for application to intracellular drug delivery. Previous work [Chakravarty *et al.,* Nat. Nanotechnol. **5**, 607–611 (2010)] has shown that cells immersed in nanoparticle-laden fluid underwent transient permeabilization when exposed to pulsed laser light. It was hypothesized that the stresses leading to cell membrane permeabilization were generated by impulsive pressures resulting from rapid nanoparticle thermal expansion. To assist in the study of the drug delivery technique, for which high uptake and viability rates have been demonstrated, an experimental method was developed for parametric assessment of photoacoustic output in the absence of field-perturbing elastic boundaries. This paper presents calibrated acoustic pressures from laser-irradiated streams, showing the impact of parameters including particle type, host liquid, and spatial distribution of laser energy.

4:45

4pBAb8. Modeling ultrasonic scattering from high-concentration cell pellet biophantoms using polydisperse structure functions. Aiguo Han and William D. O'Brien (Univ. of Illinois at Urbana-Champaign, 405 N. Mathews, Urbana, IL 61801, han51@uiuc.edu)

Backscattering coefficient (BSC) has been used extensively to characterize tissue. In most cases, sparse scatterer concentrations are assumed. However, many types of tissues have dense scattering media. This study models the scattering of dense media. Structure functions (defined herein as the total BSC divided by incoherent BSC) are used to take into account the correlation among scatterers for dense media. Structure function models are developed for polydisperse scatterers. The models are applied to cell pellet biophantoms that are constructed by placing live cells of known concentration in a mixture of bovine plasma and thrombin to form a clot. The BSCs of the biophantoms were measured using single-element transducers over 11–105 MHz. Experimental structure functions were derived by comparing the BSCs of two cell concentrations, a lower concentration (volume fraction: $<5\%$, incoherent scattering only) and a higher concentration (volume fraction: ~74%). The structure functions predicted by the models agreed with the experimental data. Fitting the models yielded cell radius estimates (Chinese hamster ovary cell: 6.9 microns, MAT cell: 7.1 microns, 4T1 cell: 8.3 microns) that were consistent with direct light microscope measures (Chinese hamster ovary: 6.7 microns, MAT: 7.3 microns, 4T1: 8.9 microns). [Work supported by NIH CA111289.]

5:00

4pBAb9. Characterizing collagen microstructure using high frequency ultrasound. Karla P. Mercado (Dept. of Biomedical Eng., Univ. of Rochester, 553 Richardson Rd., Rochester, NY 14623, karlapatricia.mercado@gmail.com), María Helguera (Ctr. for Imaging Sci., Rochester Inst. of Technol., Rochester, NY), Denise C. Hocking (Dept. of Pharmacology and Physiol., Univ. of Rochester, Rochester, NY), and Diane Dalecki (Dept. of Biomedical Eng., Univ. of Rochester, Rochester, NY)

Collagen is the most abundant extracellular matrix protein in mammals and is widely investigated as a scaffold material for tissue engineering. Collagen provides structural properties for scaffolds and, importantly, the microstructure of collagen can affect key cell behaviors such as cell migration and proliferation. This study investigated the feasibility of using high-frequency quantitative ultrasound to characterize collagen microstructure, namely, collagen fiber density and size, nondestructively. The integrated backscatter coefficient (IBC) was employed as a quantitative ultrasound parameter to characterize collagen microstructure in 3-D engineered hydrogels. To determine the relationship between the IBC and collagen fiber density, hydrogels were fabricated with different collagen concentrations (1–4 mg/mL). Further, collagen hydrogels polymerized at different temperatures (22–37°C) were investigated to determine the relationship between the IBC and collagen microfiber size. The IBC was computed from measurements of the backscattered radio-frequency data collected using a single-element transducer (38-MHz center frequency, 13–47 MHz bandwidth). Parallel studies using second harmonic generation microscopy verified changes in collagen microstructure. Results showed that the IBC increased with increasing collagen concentration and decreasing polymerization temperature. Further, we demonstrated that parametric images of the IBC were useful for assessing spatial variations in collagen microstructure within hydrogels.

5:15

4pBAb10. Surface roughness and air bubble effects on high-frequency ultrasonic measurements of tissue. Percy D. Segura, Caitlin Carter (Biology, Utah Valley Univ., 800 W. University Parkway, Orem, UT 84058-5999, pdsegura86@gmail.com), and Timothy E. Doyle (Phys., Utah Valley Univ., Orem, UT)

High-frequency (HF) ultrasound (10–100 MHz) has shown the ability to differentiate between healthy tissue, benign pathologies, and cancer in breast cancer surgical samples. It is hypothesized the sensitivity of HF ultrasound to breast cancer is due to changes in the microscopic structure of the tissue. The objective of this study was to determine the effects of surface roughness and air bubbles on ultrasound results. Since the testing is done with tissue inside a plastic bag, small air bubbles may form between the bag and tissue and interfere with test results. Data were collected on bovine and canine tissues to observe changes in HF readings in various organs and positions within specific tissues. Phantom samples were also created to mimic tissue with irregular surfaces and air bubbles. Samples were sealed into plastic bags, coupled to 50-MHz transducers using glycerin, and tested in pitch-catch and pulse-echo modes. The canine and bovine tissues produced similar results, with peak density trending with tissue heterogeneity. The surface grooves in bovine cardiac tissue also contributed to differences in peak densities. In phantom experiments, bubbles only affected peak density when they were isolated in the sample, but irregular surface structure had a strong effect on peak density.

Session 4pEA

Engineering Acoustics: Devices and Flow Noise

Roger T. Richards, Chair
US Navy, 169 Payer Ln., Mystic, CT 06355

Contributed Papers

1:30

4pEA1. Effect of fire and high temperatures on alarm signals. Mustafa Z. Abbasi, Preston S. Wilson, and Ofodike A. Ezekoye (Appl. Res. Lab. and Dept. of Mech. Eng., The Univ. of Texas at Austin, 204 E Dean Keeton St., Austin, TX 78751, mustafa_abbasi@utexas.edu)

Firefighters use an acoustic alarm to recognize and locate other firefighters that need rescue. The alarm, codified under NFPA 1982 : Standard for Personal Alert Safety System (PASS), is typically implemented in firefighter's SCBA (self-contained breathing apparatus) and is carried by a majority of firefighter in the United States. In the past, the standard specified certain frequency tones and other parameters and left implementation up to manufacturers, leading to an infinite number of possibilities that could satisfy the standard. However, there is a move to converge the standard to a single alarm sound. The research presented provides science-based guidance for the next generation of PASS signal. In the two previous ASA meetings, a number of experimental and numerical studies were presented regarding the effect of temperature stratification on room acoustics. The present work uses models developed under those studies to quantify the effect of various signal parameters (frequency ranges, time delay between successive alarms, temporal envelope etc.) on the signal heard by a firefighter. Understanding the effect of these parameters will allow us to formulate a signal more resistant to distortion caused by the fire. [Work supported by U.S. Department of Homeland Security Assistance to Firefighters Grants Program.]

1:45

4pEA2. Acoustic impedance of large orifices in thin plates. Jongguen Lee, Tongxun Yi, Katsuo Maxted, Asif Syed, and Cameron Crippa (Aerosp. Eng., Univ. of Cincinnati, 539 Lowell Ave. Apt. #3, Cincinnati, OH 45220, maxtedkj@mail.uc.edu)

Acoustic impedance of large orifices (0.5–0.75 in. diameter) in thin plates (0.062 in. thickness) was investigated. This work extended the scope previously studied by Stinson and Shaw [Stinson and Shaw, Acoust. Soc. Am. **77**, 2039 (1985)] to orifice diameters that were 32 to 584 times greater than the boundary layer thickness. For a frequency range of 0.3–2.5 kHz, the resistive and reactive components were determined from an impedance tube with six fixed microphones. Sound pressure levels (SPL) were varied from 115 to 145 dB. The transition regime from constant to increasing resistances occurred at higher frequencies for larger diameters. Resistance measurements after the transition regime were in good agreement with Thurston's theory [Thurston, J. Acoust. Soc. Am. **24**, 653–656 (1952)] coupled with Morse and Ingard's resistance factor [Morse and Ingard, *Theoretical Acoustics* (McGraw-Hill, New York, 1969)]. Measured reactances remained constant at magnitudes predicted by Thurston's theory.

2:00

4pEA3. Temperature effect on ultrasonic monitoring during a filtration procedure. Lin Lin (Eng., Univ. of Southern Maine, 37 College Ave., 131 John Mitchell Ctr., Gorham, ME 04038, llin@usm.maine.edu)

Membranes are used extensively for a wide variety of commercial separation applications including those in the water purification, pharmaceutical, and food processing industries. Fouling is a major problem associated with membrane-based liquid separation processes because it can often severely limit process performance. The use of ultrasonic monitoring technique for the characterization of membranes and membrane processes has been widely used by university researchers and industrial groups for a variety of applications including membrane fouling, compaction, formation, defect detection, and morphology characterization. However, during the industrial application, such as desalination procedure, temperature of the feed liquid is not constant. This change of the temperature brings in the concern that whether the change of the ultrasonic signal is caused by the fouling or by the temperature change. This research is focus on to verify the degree of effect of temperature to ultrasonic signal, and provide a method that calibrate the temperature effect for real applications.

2:15

4pEA4. Acoustical level measuring device. Robert H. Cameron (Eng. Technol., NMSU (Retired), 714 Winter Dr., El Paso, TX 79902-2129, rcameron@elp.rr.com)

This abstract is for a poster session to describe a patent application made to the patent office in November 2012. The patent describes a system and method for determining the level of a substance in a container, based on measurement of resonance from an acoustic circuit that includes unfilled space within the container that changes size as substance is added or removed from the container. In particular, one application of this device is to measure the unfilled space in the fuel tanks of vehicles such as cars and trucks. For over 100 years, this measurement has been done by a simple float mechanism but, because of the development of tank design for vehicles that involve irregular shapes this method is increasingly less accurate. The proposed device will overcome these limitations and should provide a much more accurate reading of the unfilled space, and therefore, the amount of fuel in the tank since the total volume of the tank is known.

2:30

4pEA5. Noise induced hearing loss mitigation via planning and engineering. Raymond W. Fischer (Noise Control Eng. Inc., 799 Middlesex Turnpike, Ste. 4B, Billerica, MA 01821, rayf@noise-control.com), Kurt Yankaskas (Code 342, Office of Naval Res., Arlington, DC), and Chris Page (Noise Control Eng. Inc., Billerica, MA)

The US Navy, through an ONR lead effort, is investigating methods and techniques to mitigate hearing loss for the crews and warfighters. Hearing protection is a viable and increasingly popular method of reducing hearing exposure for many ship crew members; however, it has limitations on comfort and low frequency effectiveness, and is often used improperly. Proper naval vessel planning, programmatic changes, and advances in noise control engineering can also have significant impacts by inherently reducing noise exposure through ship design along with the use of passive noise control treatments. These impacts go beyond hearing loss mitigation since they can improve quality of life onboard vessels and provide enhanced warfighter performance. Such approaches also can be made to work in the lower frequency range where hearing protection is not as effective. This paper describes the programmatic and noise control methods being pursued to mitigate and control noise within the US Navy and US Marine Corps. Methodologies to assess the cost impact are also discussed.

2:45

4pEA6. Enhanced sound absorption of aluminum foam by the diffuse addition of elastomeric rubbers. Elizabeth Arroyo (Dept. of Mech. Eng., Univ. of Detroit Mercy, 547 N Gulley, Dearborn Heights, MI 48127, liz.arroyo12@gmail.com), Nassif Rayess, and Jonathan Weaver (Dept. of Mech. Eng., Univ. of Detroit Mercy, Detroit, MI)

The sound absorption properties of open cell aluminum foams are understood to be significant (Ashby *et al.Metal Foams: A Design Guide*, 2000) with theoretical models presented in the literature [J. Acoust. Soc. Am. **108**, 1697–1709 (2000)]. The pores that exist in metal foams, as artifacts of the manufacturing process, are left unfilled in the vast majority of cases. Work done by the US Navy (US patent 5895726 A) involved filling the voids with phthalonitrile prepolymer, resulting in a marked increase in sound absorption and vibration damping. The work presented here involves adding small amounts of elastomeric rubbers to the metal foam, thereby coating the ligaments of the foam with a thin layer of rubber. The goal is to achieve an increase in sound absorption without the addition of cost and weight. The work involves testing aluminum foam samples of various thicknesses and pore sizes in an impedance tube, with and without the added rubber. A design of experiment model was employed to gauge the effect of the various manufacturing parameters on the sound absorption and to set the stage for a physics-based predictive model.

3:00

4pEA7. Measures for noise reduction aboard ships in times of increasing comfort demands and new regulations. Robin D. Seiler and Gerd Holbach (EBMS, Technische Unversität Berlin, Salzufer 17-19, SG 6, Berlin 10967, Germany, r.seiler@tu-berlin.de)

Through the revision of the "Code of Noise Levels on Board Ships," the International Maritime Organization has tightened its recommendations from 1984 by lowering the allowed maximum noise exposure levels on board ships. Hereby, the most significant change can be observed for cabins. To consider the effects of noise on health and comfort their noise level limits were reduced by 5 dB to 55 dB(A) equivalent continuous SPL. Another important alteration is that parts of the new code will be integrated into the SOLAS-Convention, and therefore, some of its standards will become mandatory worldwide. In order to meet the increasing demands, the focus has to be put on noise reduction measures in receiving rooms and along the sound propagation paths since the opportunity to use noise reduced devices or machines is not always given. This study gives an overview of the current noise situation on board of different types of ships. The efficiency of measures for noise reduction is discussed with focus on cabins and cabin-like receiving rooms. Especially, the role of airborne sound radiation from ship windows induced by structure-borne sound is investigated.

3:15

4pEA8. Investigation of structural intensity applied to carbon composites. Mariam Jaber, Torsten Stoewer (Structural Dynam. and Anal., BMW Group, Knorrstr. 147, München 80788, Germany, mariam.jaber@bmw.de), Joachim Bös, and Tobias Melz (System Reliability and Machine Acoust. SzM, Technische Universität Darmstadt , Darmstadt, Germany)

Structures made from carbon composite materials are rapidly replacing metallic ones in the automotive industry because of their high strength to weight ratio. The goal of this study is to enhance acoustic comfort of cars made from carbon composites by comparing various carbon composites in order to find the most suitable composite in terms of mechanical and dynamic properties. In order to achieve this goal, the structural intensity method was implemented. This method can give information concerning the path of energy propagated through structures and the localization of vibration sources and sinks. The significance of the present research is that it takes into account the effect of the material damping on the dissipation of the energy in a structure. The damping of the composite is presented as a function of its micro and macro mechanical properties, frequency, geometry, and boundary conditions. The damping values were calculated by a 2D analytical multi-scale model based on the laminate theory. The benefit of this research for acoustics is that it demonstrates the effect of material properties on passive control. Consequently, structural energy propagated in carbon composite structures will be reduced and less noise will be radiated.

3:30

4pEA9. Experimental research on acoustic agglomeration of fine aerosol particles in the standing-wave tube with abrupt section. Zhao Yun, Zeng Xinwu, and Gong Changchao (Optical-Electron. Sci. and Eng., National University of Defense Technol., Changsha 410073, China, zhaoyun@nudt.edu.cn)

There is great concern about air pollution caused by fine aerosol particles, which are difficult to be removed by conventional removal system. Acoustic agglomeration is proved to be a promising method for particle control by coagulating the small particles into larger ones. Removal efficiency was grown rapidly as acoustic intensity increased. A standing-wave tube system with abrupt section was designed and built up to generate high intensity sound waves above 160 dB and avoid strong shock waves. Extensive tests were carried out to investigate the acoustic field and removal characteristics of coal-fired inhalation particles. For the development of industrial level system, a high power air-modulated speaker was applied and an insulation plate was used to separate flow induced sound. Separate experiments to determine the difference of plane standing-wave field and high order mode were conducted. The experimental study has demonstrated that agglomeration increases as sound pressure level, mass loading, and exposure time increase. The optimal frequency is around 2400 Hz for attaining integral removal effectiveness. The agglomeration rate is larger (above 86%) as much greater sound level is achieved for the pneumatic source and high order mode. The mechanism and testing system can be used effectively in industrial processes.

3:45–4:00 Break

4:00

4pEA10. Aerodynamic and acoustic analysis of an industrial fan. Jeremy Bain (Bain Aero LLC, Stockbridge, GA), Gang Wang (Ingersoll Rand, La Crosse, Wisconsin), Yi Liu (Ingersoll Rand, 800 Beaty St., Davidson, North Carolina 28036, yiliu@irco.com), 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 and increasing computational power, 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 directly predicted using an Overset grid system and high order finite difference schemes with acceptable fidelity.

4:15

4pEA11. On the acoustic and aerodynamic performance of serrated airfoils. Xiao Liu (Mech. Eng., Univ. of Bristol, Bristol, United Kingdom), Mahdi Azarpeyvand (Mech. Eng. Dept., Univ. of Bristol, Bristol BS8 1TR, United Kingdom, m.azarpeyvand@bristol.ac.uk), and Phillip Joseph (Inst. of Sound and Vib. Res., Univ. of Southampton, Southampton, United Kingdom)

This paper is concerned with the aerodynamic and aeroacoustic performance of airfoils with serrated trailing edges. Although a great deal of research has been directed toward the application of serrations for reducing the trailing-edge noise, the aerodynamic performance of such airfoils has received very little research attention. Sawtooth and slitted-sawtooth trailing edges with specific geometrical characteristics have been shown to be effective in reducing the trailing edge noise over a wide range of frequencies. It has, however, also been shown that they can alter the flow characteristics near the trailing edge, namely the boundary layer thickness and surface-pressure fluctuations, and the wake formation. To better understand the effects of serrations, we shall carry out various acoustic and wind tunnel tests for a NACA6512-10 airfoil with various sawtooth, slitted and slitted-sawtooth trailing edge profiles. Flow measurements are carried out using PIV, LDV and hot-wire anemometry and the steady and unsteady forces on the airfoil are obtained using a three-component force balance system.

Results are presented for a wide range of Reynolds numbers and angles of attack. The results have shown that the use of sharp serrations can significantly change the aerodynamic performance and wake characteristics of the airfoil.

4:30

4pEA12. An experimental investigation on the near-field turbulence for an airfoil with trailing-edge serrations at different angles of attack. Kunbo Xu and Weiyang Qiao (School of Power and Energy, Northwestern PolyTech. Univ., No.127 Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, 364398100@qq.com)

The ability to fly silently of most owl species has long been a source of inspiration for finding solutions for quieter aircraft and turbo machinery. This study concerns the mechanisms of the turbulent broadband noise reduction for an airfoil with the trailing edge serrations while the angles of attack varies from $+5^\circ$ to Ő5°. The turbulence spatio-temporal information are measured with 3D hot-wire. The experiment is carried out in the Northwestern Polytechnical University low speed open jet wind tunnel on the SD2030 airfoil. λ/h = 0.2. It is showed the spreading rate of the wake and the decay rate of the wake centerline velocity deficit increased with serrated edge compared to the straight edge, and the three components of velocity changed differently with serrated trailing edge while the angle of attack was changed. It is also found that the turbulence peak occurs further from the airfoil surface in the presence of the serrations, and the serrations widened the mix area which allowed the flow mixed together ahead of the schedule.

4:45

4pEA13. An experimental investigation on the near-field turbulence and noise for an airfoil with trailing-edge serrations. Kunbo Xu (School of Power and Energy, Northwestern Polytechnical Univ., No.127 Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, 364398100@qq.com)

This study concerns the mechanisms of the turbulent broadband noise reduction for an airfoil with the trailing edge serrations. The turbulence spatio-temporal information were measured with 3D hot-wire and the noise results were acquired with a line array. The experiment is carried out in the Northwestern Polytechnical University low speed open jet wind tunnel on the SD2030 airfoil. λ/h = 0.2. It is showed the spreading rate of the wake and the decay rate of the wake centerline velocity deficit increased with serrated edge compared to the straight edge, shedding vortex peaks appeared in the wake, and the three components of velocity changed differently with serrated trailing edge. Serrated trailing edge structure could reduce the radiated noise was proofed by noise results.

THURSDAY AFTERNOON, 8 MAY 2014 BALLROOM C, 2:00 P.M. TO 4:30 P.M.

Session 4pMUa

Musical Acoustics: Automatic Musical Accompaniment Systems

Christopher Raphael, Cochair
Indiana Univ., School of Informatics and Computing, Bloomington, IN 47408

James W. Beauchamp, Cochair
Music and Electrical and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1002 Eliot Dr., Urbana, IL 61801-6824

Invited Papers

2:00

4pMUa1. Human-computer music performance: A brief history and future prospects. Roger B. Dannenberg (School of Comput. Sci., Carnegie Mellon Univ., 5000 Forbes Ave., Pittsburgh, PA 15213, rbd@cs.cmu.edu)

Computer accompaniment began in the eighties as a technology to synchronize computers to live musicians by sensing, following, and adapting to expressive musical performances. The technology has progressed from systems where performances were modeled as sequences of discrete symbols, i.e., pitches, to modern systems that use continuous probabilistic models. Although score following techniques have been a common focus, computer accompaniment research has addressed many other interesting topics, including the musical adjustment of tempo, the problem of following an ensemble of musicians, and making systems more robust to unexpected mistakes by performers. Looking toward the future, we find that score following is only one of many ways musicians use to synchronize. Score following is appropriate when scores exist and describe the performance accurately, and where timing deviations are to be followed rather than ignored. In many cases, however, especially in popular music forms, tempo is rather steady, and performers improvise many of their parts. Traditional computer accompaniment techniques do not solve these important music performance scenarios. The term Human-Computer Music Performance (HCMP) has been introduced to cover a broader spectrum of problems and technologies where humans and computers perform music together, adding interesting new problems and directions for future research.

2:25

4pMUa2. The cyber-physical system approach for automatic music accompaniment in Antescofo. Arshia Cont (STMS 9912-CNRS, UPMC, Inria MuTant Team-Project, IRCAM, 1 Pl. Igor Stravinsky, Paris 75004, France, arshia.cont@ircam.fr), José Echeveste (STMS 9912, IRCAM, CNRS, Inria MuTant Team-Project, Sorbonne Univ., UPMC Paris 06, Paris, France), and Jean-Louis Giavitto (IRCAM, UPMC, Inria MuTant team-project, CNRS STMS 9912, Paris, France)

A system capable of undertaking automatic musical accompaniment with human musicians should be minimally able to undertake real-time listening of incoming music signals from human musicians, and synchronize its own actions in real-time with that of musicians according to a music score. To this, one must also add the following requirements to assure correctness: Fault-tolerance to human or machine listening errors, and best-effort (in contrast to optimal) strategies for synchronizing heterogeneous flows of information. Our approach in Antescofo consists of a tight coupling of real-time Machine Listening and Reactive and Timed-Synchronous systems. The machine listening in Antescofo is in charge of encoding the dynamics of the outside environment (i.e., musicians) in terms of incoming events, tempo and other parameters from incoming polyphonic audio signal; whereas the synchronous timed and reactive component is in charge of assuring correctness of generated accompaniment. The novelty in Antescofo approach lies in its focus on Time as a semantic property tied to correctness rather than a performance metric. Creating automatic accompaniment out of symbolic (MIDI) or audio data follows the same procedure, with explicit attributes for synchronization and fault-tolerance strategies in the language that might vary between different styles of music. In this sense, Antescofo is a cyber-physical system featuring a tight integration of, and coordination between heterogeneous systems including human musicians in the loop of computing.

2:50

4pMUa3. Automatic music accompaniment allowing errors and arbitrary repeats and jumps. Shigeki Sagayama (Div. of Information Principles Res., National Inst. of Informatics, 2-1-2, Hitotsubashi, Choyoda-ku, Tokyo 101-8430, Japan, sagayama@nii.ac.jp), Tomohiko Nakamura (Graduate School of Information Sci. and Technol., Univ. of Tokyo, Tokyo, Japan), Eita Nakamura (Div. of Information Principles Res., National Inst. of Informatics, Japan, Tokyo, Japan), Yasuyuki Saito (Dept. of Information Eng., Kisarazu National College of Technol., Kisarazu, Japan), and Hirokazu Kameoka (Graduate School of Information Sci. and Technol., Univ. of Tokyo, Tokyo, Japan)

Automatic music accompaniment is considered to be particularly useful in exercises, rehearsals and personal enjoyment of concerto, chamber music, four-hand piano pieces, and left/right hand filled in to one-hand performances. As amateur musicians may make errors and want to correct them, or he/she may want to skip hard parts in the score, the system should allow errors as well as arbitrary repeats and jumps. Detecting such repeats/jumps, however, involves a large complexity of search for maximum likelihood transition from one onset timing to another in the entire score for every input event. We have developed several efficient algorithms to cope with this problem under practical assumptions used in an online automatic accompaniment system named "Eurydice." In Eurydice for MIDI piano, the score of music piece is modeled by Hidden Markov Model (HMM) as we proposed for rhythm modeling in 1999 and the maximum likelihood score following is done to the polyphonic MIDI input to yield the accompanying MIDI output (e.g., orchestra sound). Another version of Eurydice accepts monaural audio signal input and accompanies to it. Trills, grace notes, arpeggio, and other issues are also discussed. Our video examples include concertos with MIDI piano and piano accompanied sonatas for acoustic clarinet.

3:15

4pMUa4. The informatics philharmonic. Christopher Raphael (Comput. Sci., Indiana Univ., School of Informatics and Computing, Bloomington, IN 47408, craphael@indiana.edu)

I present ongoing work in developing a system that accompanies a live musician in a classical concerto-type setting, providing a flexible ensemble the follows the soloist in real-time and adapts to the soloist's interpretation through rehearsal. An accompanist must hear the soloist. The program models hearing through a hidden Markov model that can accurately and reliably parse highly complex audio in both offline and online fashion. The probabilistic formulation allows the program to navigate the latency/accuracy tradeoff in online following, so that onset detections occur with greater latency (and greater latency) when local ambiguities arise. For music with a sense of pulse, coordination between parts must be achieved by anticipating future evolution. The program develops a probabilistic model for musical timing, a Bayesian Belief Network, that allows the program to anticipate where future note onsets will occur, and to achieve better prediction using rehearsal data. The talk will include a live demonstration of the system on a staple from the violin concerto repertoire, as well as applications to more forward-looking interactions between soloist and computer controlled instruments.

3:40

4pMUa5. Interactive conducting systems overview and assessment. Teresa M. Nakra (Music, The College of New Jersey, P.O. Box 7718, Ewing, NJ 08628, nakra@tcnj.edu)

"Interactive Conducting" might be defined as the accompaniment of free gestures with sound—frequently, but not necessarily, the sounds of an orchestra. Such systems have been in development for many decades now, beginning with Max Mathews' "Daton" interface and "Conductor" program, evolving to more recent video games and amusement park experiences. The author will review historical developments in this area and present several of her own recent interactive conducting projects, including museum exhibits, simulation/training systems for music students, and data collection/analysis methods for the study of professional musical behavior and response. A framework for assessing and evaluating effective characteristics of these systems will be proposed, focusing on the reactions and experiences of users/subjects and audiences.

4:05

4pMUa6. The songsmith story, or how a small-town hidden Markov model dade it to the big time. Sumit Basu, Dan Morris, and Ian Simon (Microsoft Res., One Microsoft Way, Redmond, WA 98052, sumitb@microsoft.com)

It all started with a simple idea—that perhaps lead sheets could be predicted from melodies, at least within a few options for each bar. Early experiments with conventional models led to compelling results, and by designing some user interactions along with an augmented model, we were able to create a potent tool with a range of options, from an automated backing band for musical novices to a flexible musical scratchpad for songwriters. The academic papers on the method and tool led to an unexpected level of external interest, so we decided to make a product for consumers, thus was Songsmith born. What came next surprised us all—from internet parodies to stock market melodies to over 600 000 downloads and a second life in music education, Songsmith has been an amazing lesson in what happens when research and the real world collide, sometimes with unintended consequences. In this talk, I'll take you through our story, from the technical beginnings to the Internet-sized spectacle to the vast opportunities in future work, sharing with you the laughter, the heartbreak, the tears, and the joy of bringing Songsmith to the world.

THURSDAY AFTERNOON, 8 MAY 2014 BALLROOM C, 4:45 P.M. TO 6:00 P.M.

Session 4pMUb

Musical Acoustics: Automatic Accompaniment Demonstration Concert

Christopher Raphael, Cochair
Indiana Univ., School of Informatics and Computing, Bloomington, IN 47408

James W. Beauchamp, Cochair
Music and Electrical and Comput. Eng., Univ. of Illinois at Urbana-Champaign, 1002 Eliot Dr., Urbana, IL 61801-6824

Music performed by Christopher Raphael (oboe), Roger Dannenberg (trumpet), accompanied by their automatic systems.

THURSDAY AFTERNOON, 8 MAY 2014 557, 1:30 P.M. TO 5:10 P.M.

Session 4pNS

Noise: Out on a Limb and Other Topics in Noise

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

Invited Papers

1:30

4pNS1. Necessity as the mother of innovation: Adapting noise control practice to very different set of mechanical system design approaches in an age of low energy designs. Scott D. Pfeiffer (Threshold Acoust. LLC, 53 West Jackson Blvd., Ste. 815, Chicago, IL 60604, spfeiffer@thresholdacoustics.com)

The shift in Mechanical Systems design to natural ventilation, dedicated outside air systems, variable refrigerant flow, and the return to radiant systems all present new challenges in low-noise systems. Case studies of current projects explore the sound isolation impact of natural ventilation, the benefits of reduced air quantity in dedicated outside air, the distributed noise issues in variable refrigerant flow, and the limitations of radiant systems as they apply in performing arts and noise critical spaces.

1:50

4pNS2. Readily available noise control for residences in Boston. Nancy S. Timmerman (Nancy S. Timmerman, P.E., 25 Upton St., Boston, MA 02118, nstpe@hotmail.com)

Urban residential noise control may involve high-end interior finishes, insufficient noise reduction between neighbors (in a same building), or interior/exterior noise reduction for mechanical equipment or transportation where the distances are small or non-existent. Three residences in Boston's South End, where the author is a consultant (and resident), will be discussed. The area consists of brownstones built in the mid-nineteenth century, with granite foundations, masonry facades, and common brick walls. Treatments were used which were acceptable to the "users"—neighbors on both sides of the fence.

Contributed Papers

2:10

4pNS3. Singing in the wind; noise from railings on coastal and high-rise residential construction. Kenneth Cunefare (Arpeggio Acoust. Consulting, LLC, Mech. Eng., Atlanta, GA 30332-0405, ken.cunefare@me.gatech.edu)

Beach-front and high-rise residential buildings are commonly exposed to sustained high winds. Balcony railings with long spans and identical pickets on uniform spacing may be driven into extremely high amplitude synchronous motion due to phase and frequency locked vortex shedding. The railing motion can excite structural vibration in floor slabs which can propagate into units and produce undesirable tone-rich noise within the units, noise that stands out well above the wind noise that also propagates into the units. Solution of this problem requires breaking the physical phenomena that induce the railing motion, including blanking off the railings; stiffening the railings; and breaking the symmetry of the individual pickets. The problem may be further complicated by questions of who should pay for the remediation of the problem, and the costs associated with remediating numerous units, particularly on high-rise developments. Increased awareness during the design phase of the potential for this problem may reduce the need for post-construction controls.

2:25

4pNS4. What do teachers think about noise in the classroom? Ana M. Jaramillo (Ahnert Feistel Media Group, 3711 Lake Dr., 55422, Robbinsdale, MN 55422, ana.jaramillo@afmg.eu), Michael G. Ermann, and Patrick Miller (School of Architecture + Design, Virginia Tech, Blacksburg, VA)

Surveys were sent to 396 Orlando-area elementary school teachers to gauge their subjective evaluation of noise in their classroom, and their general attitudes toward classroom noise. The 87 responses were correlated with the types of mechanical systems in their respective schools: (1) fan and compressor in room, (2) fan in room and remote compressor, or (3) remote fan and remote compressor. Results were also compared to the results of a previous study of the same 73 schools that linked school mechanical system type with student achievement. While teachers were more likely to be annoyed by noise in the schools with the noisiest types of mechanical systems, they were still less likely to be annoyed than the research might suggest—and when teachers did express annoyance, it was more likely to be centered around the kind of distracting noise generated by other children in adjacent corridors than by mechanical system noise.

2:40

4pNS5. Sound classification of dwellings—A comparison between national schemes in Europe and United States. Umberto Berardi (Civil and Environ. Eng. Dept., Worcester Polytechnic Inst., via Orabona 4, Bari 70125, Italy, u.berardi@poliba.it)

Schemes for the classification of dwellings related to different performances have been proposed in the last years worldwide. The general idea behind previous schemes relates to the increase in the real estate value that should follow a label corresponding to a better performance. In particular, focusing on sound insulation, national schemes for acoustic classification of dwellings have been developed in more than ten European countries. These schemes define classification classes according to different levels of sound insulation. The considered criteria are the airborne and impact sound insulation between dwellings, the facade sound insulation, and the equipment noise. Originally, due to the lack of coordination among European countries, a significant diversity among the schemes occurred; the descriptors, number of classes, and class intervals varied among schemes. However, in the last year, an "acoustic classification scheme for dwellings" has been proposed within a ISO technical committee. This paper compares existing classification schemes with the current situation in the United States. The hope is that by increasing cross-country comparisons of sound classification schemes, it may be easier to exchange experiences about constructions fulfilling different classes and by doing this, reduce trade barriers, and increase the sound insulation of dwellings.

2:55

4pNS6. Sound insulation analysis of residential building at China. Zhu Xiangdong, Wang Jianghua, Xue Xiaoyan, and Wang Xuguang (The Bldg. Acoust. Lab of Tsinghua Univ., No. 104 Main Academic Bldg. Architectural Physical Lab., Tsinghua Univ., Beijing, Beijing 100084, China, zxd@abcd.edu.cn)

Residential acoustic environment is one of the living environments that are most closely related to the daily life. The high-quality residential acoustic environment depends not only on the urban planning, building design, construction, and supervision, but also on the related regulations. In some developed countries, the residential acoustic regulations have been built up and evolved into a relatively complete system with high quality standards required. This thesis (1) conducted a questionnaire survey for resident building which be constructed at different period; (2) investigate the Technical level, the legal system, and the quality of residents to analysis the sound environment satisfaction of resident and compare it with developed countries.

3:10–3:25 Break

3:25

4pNS7. Relationship between air infiltration and acoustic leakage of building enclosures. Ralph T. Muehleisen, Eric Tatara, and Brett Bethke (Decision and Information Sci., Argonne National Lab., 9700 S. Cass Ave., Bldg. 221, Lemont, IL 60439, rmuehleisen@anl.gov)

Air infiltration, the uncontrolled leakage of air into buildings through the enclosure from pressure differences across it, accounts for a significant fraction of the heating energy in cold weather climates. Measurement and control of this infiltration is a necessary part of reducing the energy and carbon footprint of both current and newly constructed buildings. The most popular method of measuring infiltration, whole building pressurization, is limited to small buildings with fully constructed enclosures, which makes it an impractical method for measuring infiltration on medium to large buildings or small buildings still under construction. Acoustic methods, which allow for the measurement of infiltration of building sections and incomplete enclosures, have been proposed as an alternative to whole building pressurization. These new methods show great promise in extending infiltration measurement to many more buildings, but links between the acoustic leakage characteristics and the infiltration characteristics of typical enclosures are required. In this paper, the relationship between the acoustic leakage and the air infiltration through typical building envelope cracks is investigated. [This work was supported by the U.S. Department of Energy under Contract No. DE-AC02-06CH11357.]

3:40

4pNS8. Hemi-anechoic chamber qualification and comparison of room qualification standards. Madeline A. Davidson (Acoust. and Mech., Trane Lab., 700 College Dr. SPO 542, Luther College, Decorah, Iowa 52101, davima07@luther.edu)

The hemi-anechoic chamber at the Trane Laboratory in La Crosse, Wisconsin, is commonly used for acoustic testing of machinery and equipment. As required by standards, it must periodically be qualified. Sound measurements taken in a hemi-anechoic facility often depend on the assumption that the chamber is essentially free-field. To verify that the room is sufficiently anechoic, the procedures in ANSI/ASA Standard S12.55-2012/ISO 3745:2012 and ISO Standard 26101-2012 are followed. One challenge of a room qualification is finding adequate sound sources. Sources used in the qualification procedure must be Omni-directional, so directionality measurements must be taken to prove that a source is suitable for the room qualification procedure. The specific qualification procedure described in this paper involved two sound sources—a compression driver and a 6 in. $\times$ 9 in. speaker. In addition, the particular method described in this paper involves a temporary plywood floor and six microphone traverse paths extending out from the center of the chamber. This approach to qualifying a facility is

expected to define what part of the room is adequately anechoic. This paper will describe the results obtained when following each of these standards.

3:55

4pNS9. Improvement of the measurement of the sound absorption using the reverberation chamber method. Martijn Vercammen (Peutz, Lindenlaan 41, Mook 6585 ZH, Netherlands, m.vercammen@peutz.nl) and Margriet Lautenbach (Peutz, Zoetermeer, Netherlands)

The random incidence absorption coefficient is measured in a reverberation room according to ISO 354 or ASTM C423-09a. It is known that the inter laboratory accuracy under Reproducibility conditions of these results is still not very well. It is generally assumed that the limited diffusion properties of reverberation rooms, especially with a strongly sound absorbing sample, are the main reason for the bad reproducibility values for the sound absorption between laboratories. Reverberation rooms should be made much more diffuse to reduce the interlaboratory differences. However there are practical limitations in quantifying and improving the diffuse field conditions. The measured sound absorption still seems to be the most sensitive descriptor of the diffuse field conditions. A way to further reduce the interlaboratory differences is the use of a reference absorber to qualify a room and to calibrate the results of a sound absorption measurement. In the presentation an overview will be given of the research performed and some suggestions for the new version of ISO 354 will be given.

4:10

4pNS10. When acoustically rated doors fail to perform as rated, who is responsible—Manufacturer or installer? Marlund E. Hale (Adv. Eng. Acoust., 663 Bristol Ave., Simi Valley, CA 93065, mehale@aol.com)

Acoustical doors are designed, manufactured, and sold by several companies in the United States. They are available in multiple styles and acoustical performance ratings. The doors are specified, selected, and purchased based on the published performance ratings provided by the manufacturers, which often have had their doors tested by NAVLAP accredited acoustical testing laboratories. Of course, it should be understood by the acoustical door specifier that lab-rated doors will rarely, if ever, perform as rated after field installation. This paper presents field performance test results for numerous acoustical doors that significantly failed even the lower expected field performance criteria. The acoustical doors were all tested in-situ after they were installed in several different venues by the manufacturer's or vendor's trained and/or certified acoustical door installers. Reasons for certain field-performance failures are discussed and specific remedies are recommended.

4:25

4pNS11. Sound absorption of parallel arrangement of multiple microperforated panel absorbers at oblique incidence. Chunqi Wang, Lixi Huang, Yumin Zhang (Lab of AeroDynam, and Acoust., Zhejiang Inst. of Res. and Innovation and Dept of Mech. Eng., The Univ. of Hong Kong , Pokfulam Rd., Hong Kong, lixi@hku.hk)

Many efforts have been made to enhance the sound absorption performance of micro-perforated panel (MPP) absorbers. Among them, one straightforward approach is to arrange multiple MPP absorbers of different frequency characteristics in parallel so as to combine different frequency bands together, hence an MPP absorber array. In previous study, the parallel absorption mechanism is identified to be contributed by three factors: (i) the strong local resonance absorption, (ii) the supplementary absorption by nonresonating absorbers, and (iii) the change of environmental impedance conditions; and the local resonance absorption mechanism accounts for the increased equivalent acoustic resistance of the MPP. This study seeks to examine how the MPP absorber array performs at oblique incidence and in diffuse field. One major concern here is how the incidence angle of the sound waves affects the parallel absorption mechanism. In this study, a finite element model is developed to simulate the acoustic performance of an infinitely large MPP absorber array. Numerical results show that the sound absorption coefficients of the MPP absorber array may change noticeably as the incidence angle varies. The diffuse field sound absorption coefficients of a prototype specimen are measured in a reverberation room and compared with the numerical predictions.

4:40

4pNS12. Reverberation time in ordinary rooms of typical residences in Southern Brazil. Michael Ã. Klein, Andriele da Silva Panosso, and Stephan Paul (DECC-CT-UFSM, UFSM, Av. Roraima 1000, Camobi, Santa Maria 97105-900, Brazil, michaelklein92@hotmail.com)

In order to develop a subjective evaluation to assess the annoyance related to impact noise, it is necessary to record samples of sounds in an impact chamber that is acoustically representative for ordinary rooms, especially with respect to reverberation time. To define the target reverberation time measurements were carried out in 30 typical residences in Southern Brazil. This study presents the characteristic reverberation times of 30 furnished living rooms and 30 furnished bedrooms in buildings and houses with an average age of 34 years, 40% of them with wooden floor coverings, not as usual in modern constructions. The median T30 at 1 kHz for living rooms with an average volume of 63.60m^3 (std dev: 18.27m^3) was 0.68 s (std dev: 0.14 s), thus higher than the reference TR = 0.5 s according to EN ISO 140 parts 4, 5, and 7. The median T30 at 1 kHz for bedrooms with average volume of 33.76m^3 (std dev: 8.38m^3) was 0.49 s (std dev: 0.13 s), nearly exact the reference TR according to EN ISO 140 parts 4, 5, and 7. Data will also be compared to studies from other countries.

4:55

4pNS13. Research on the flow resistance of acoustic materials—Takes Concert Hall at Gulangyu Music School in Xiamen as an Example. Peng Wang, Xiang Yan, Lu W. Shuai, Gang Song, and Yan Liang (Acoust. Lab., School of Architecture, Tsinghua Univ., Beijng, China, 29580150@qq.com)

Different kinds of acoustic materials are used in a concert hall design, which has different functions such as diffusing, reflecting, or absorbing. The cushion of chairs in concert halls usually uses porous sound-absorbing material, whose absorbing attributes are mainly determined by its flow resistance. In the design of Concert Hall at Gulangyu Music School in Xiamen, we measured the flow resistance of materials, trying to acquire the best sound-absorbing attributes by adjusting the flow resistance, and also tested the material samples' absorbing coefficients in reverberation room. In a nutshell, measuring and analyzing flow resistance is an advanced method in acoustic design, which could help acousticians decide the most suitable absorbing attributes of chairs, and acquire the best sound quality.

Session 4pPA

Physical Acoustics: Topics in Wave Propagation and Noise

Richard Raspet, Chair
NCPA, Univ. of Mississippi, University, MS 38677

Contributed Papers

1:00

4pPA1. Mechanisms for wind noise reduction by a spherical wind screen. Richard Raspet, Jeremy Webster, and Vahid Naderyan (National Ctr. for Physical Acoust., Univ. of MS, 1 Coliseum Dr., University, MS 38606, raspet@olemiss.edu)

Spherical wind screens provide wind noise reduction at frequencies which correspond to turbulence scales much larger than the wind screen. A popular theory is that reduction corresponds to averaging the steady flow pressure distribution over the surface. Since the steady flow pressure distribution is positive on the front of the sphere and negative on the back of the sphere, the averaging results in a reduction in measured wind noise in comparison to an unscreened microphone. A specially constructed 180 mm diameter foam sphere allows the placement of an array of probe microphone tubes just under the surface of the foam sphere. The longitudinal and transverse correlation lengths as a function of frequency and the rms pressure fluctuation distribution over the sphere surface can be determined from these measurements. The measurements show that the wind noise correlation lengths are much shorter than the correlations measured in the free stream. The correlation length weighted pressure squared average over the surface is a good predictor of the wind noise measured at the center of the wind screen. [This work was supported by the Army Research Laboratory under Cooperative Agreement W911NF-13-2-0021.]

1:15

4pPA2. Infrasonic wind noise in a pine forest; convection velocity. Richard Raspet and Jeremy Webster (National Ctr. for Physical Acoust., Univ. of MS, 1 Coliseum Dr., University, MS 38606, raspet@olemiss.edu)

Simultaneous measurements of the infrasonic wind noise, the wind velocity profile in and above the canopy, and the wind turbulence spectrum in a pine forest have been completed. The wind noise spectrum can be computed from the meteorological measurements with the assumption that the lowest frequency wind noise is generated by the turbulence field above the canopy and that the higher frequencies are generated by the turbulence within the tree layer [JASA **134**(5), 4160 (2013)]. To confirm the source region identification, an array of infrasound sensors is deployed along the approximate flow direction so that the convection velocity as a function of frequency band can be determined. This paper reports on the results of this experiment. [Work supported by the U. S. Army Research Office under grant W911NF-12-0547.]

1:30

4pPA3. The effective sound speed approximation and its implications for en-route propagation. Victor Sparrow, Kieran Poulain, and Rachel Romond (Grad. Prog. Acoust., Penn State, 201 Appl. Sci. Bldg., University Park, PA 16802, vws1@psu.edu)

The effective sound speed approximation is widely used in underwater and outdoor sound propagation using common models such as ray tracing, the parabolic equation, and wavenumber integration methods such as the fast field program. It is also used in popular specialized propagation methods such as NORD2000 and the Hybrid Propagation Model (HPM). Long ago when the effective sound speed approximation was first introduced, its shortcomings were understood. But over the years, a common knowledge of those shortcomings has waned. The purpose of this talk is to remind everyone that for certain situations the effective sound speed approximation is not appropriate. One of those instances is for the propagation of sound from aircraft cruising at en-route altitudes when wind is present. This is one situation where the effective sound speed approximation can lead to substantially incorrect sound level predictions on the ground. [Work supported by the FAA. The opinions, conclusions, and recommendations in this material are those of the authors and do not necessarily reflect the views of FAA Center of Excellence sponsoring organizations.]

1:45

4pPA4. Nonlinearity spectral analysis of high-power military jet aircraft waveforms. Kent L. Gee, Tracianne B. Neilsen, Brent O. Reichman, Derek C. Thomas (Dept. of Phys. and Astronomy, Brigham Young Univ., N243 ESC, Provo, UT 84602, kentgee@byu.edu), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

One of the methods for analyzing noise waveforms for nonlinear propagation effects is a spectrally-based nonlinearity indicator that involves the cross spectrum between the pressure waveform and square of the pressure. This quantity, which stems directly from ensemble averaging the generalized Burgers equation, is proportional to the local rate of change of the power spectrum due to nonlinearity [Morfey and Howell, AIAA J. **19**, 986–992 (1981)], i.e., it quantifies the parametric sum and difference-frequency generation during propagation. In jet noise investigations, the quadspectral indicator has been used to complement power spectral analysis to interpret mid-field propagation effects [Gee *et al*., AIP Conf. Proc. **1474**, 307–310 (2012)]. In this paper, various normalizations of the quadspectral indicator are applied to F-22A Raptor data at different engine powers. Particular attention is paid to the broadband spectral energy transfer around the spatial region of maximum overall sound pressure level. [Work supported by ONR.]

2:00

4pPA5. Evolution of the derivative skewness for high-amplitude sound propagation. Brent O. Reichman (Brigham Young Univ., 453 E 1980 N, #B, Provo, UT 84604, brent.reichman@byu.edu), Michael B. Muhlestein (Brigham Young Univ., Austin, Texas), Kent L. Gee, Tracianne B. Neilsen, and Derek C. Thomas (Brigham Young Univ., Provo, UT)

The skewness of the first time derivative of a pressure waveform has been used as an indicator of shocks and nonlinearity in both rocket and jet noise data [e.g., Gee *et al*., J. Acoust. Soc. Am. **133**, EL88–EL93 (2013)]. The skewness is the third central moment of the probability density function and demonstrates asymmetry of the distribution, e.g., a positive skewness may indicate large, infrequently occurring values in the data. In the case of nonlinearly propagating noise, a positive derivative skewness signifies occasional instances of large positive slope and more instances of negative slope as shocks form [Shepherd *et al*., J. Acoust. Soc. Am. **130**, EL8–EL13 (2011)]. In this paper, the evolution of the derivative skewness, and its interpretation, is considered analytically using key solutions of the Burgers equation. This paper complements a study by Muhlestein *et al.* [J. Acoust. Soc. Am. **134**, 3981 (2013)] that used similar methods but with a different metric.

An analysis is performed to investigate the effect of a finite sampling frequency and additive noise. Plane-wave tube experiments and numerical simulations are used to verify the analytic solutions and investigate derivative skewness in random noise waveforms. [Work supported by ONR.]

2:15

4pPA6. Application of time reversal analysis to military jet aircraft noise. Blaine M. Harker (Dept. of Phys. and Astronomy, Brigham Young Univ., N283 ESC, Provo, UT 84602, blaineharker@byu.net), Brian E. Anderson (Geophys. Group (EES-17), Los Alamos National Lab., Los Alamos, NM), Kent L. Gee, Tracianne B. Neilsen (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

The source mechanisms of jet noise are not fully understood and different analysis methods can provide insight. Time reversal (TR) is a robust data processing method that has been used in myriad contexts to localize and characterize sources from measured data, but has not extensively been applied to jet noise. It is applied here in the context of an installed full-scale military jet engine. Recently, measurements of an F-22A were taken using linear and planar microphone arrays at various engine conditions near the jet plume [Wall *et al.*, Noise Control Eng. J. **60**, 421–434 (2012)]. TR provides source imaging information as broadband and narrowband jet noise recordings are reversed and back propagated to the source region. These reconstruction estimates provide information on dominant source regions as a function of frequency and highlight directional features attributed to large-scale structures in the downstream jet direction. They also highlight the utility of TR analysis as being complementary to beamforming and other array methods. [Work supported by ONR.]

2:30–2:45 Break

2:45

4pPA7. Spectral variations near a high-performance military aircraft. Tracianne B. Neilsen, Kent L. Gee (Brigham Young Univ., N311 ESC, Provo, UT 84602, tbn@byu.edu), and Michael M. James (Blue Ridge Res. and Consulting, LLC, Asheville, NC)

Spectral characteristics of jet noise depend upon location relative to the nozzle axis. Studies of the spectral variation in the far field led to a two-source model of jet noise, in which fine-scale turbulent structures are primarily responsible for noise radiation to the nozzle sideline and large-scale turbulent structures produce the broad, dominant radiation lobe farther aft. Detailed noise measurements near an F-22A Raptor shed additional insights into this variation. An initial study [Neilsen *et al.*, J. Acoust. Soc. Am. **133**, 2116–2125] was performed with ground-based microphones in the mid-field. The similarity spectra associated with the large and fine-scale turbulent structures [Tam *et al.*, AIAA paper 96–1716 (1996)] provide a reasonable representation of measured spectra at many locations. However, there are additional features that need further investigation. This paper explores the presence of a double peak in the spectra in the maximum radiation direction and a significant change in spectral shape at the farthest aft angles using data from large measurement planes (2 m $\times$ 23 m) located 4–6 jet nozzle diameters from the shear layer. The spatial variation of the spectra provides additional insight into ties between the similarity spectra and full-scale jet noise. [Work supported by ONR.]

3:00

4pPA8. Large eddy simulation of surface pressure fluctuations generated by elevated gusts. Jericho Cain (National Ctr. for Physical Acoust., Univ. of MS, 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, Maryland 20783, jericho.cain.ctr@mail.mil), Richard Raspet (National Ctr. for Physical Acoust., Univ. of MS, University, MS), and Martin Otte (Environ. Protection Agency, Atmospheric Modeling and Anal. Div., Res. Triangle Park, NC)

A surface monitoring system that can detect turbulence aloft would benefit wind turbine damage prevention, aircraft safety, and would be a new probe to study the atmospheric boundary layer. Previous research indicated that elevated velocity events may trigger pressure fluctuations on the ground. If that is true, it should be possible to monitor elevated wind gusts by measuring these pressure fluctuations. The goal of this project was to develop a ground based detection method that monitors pressure fluctuations on the ground for indicators that a gust event may be taking place at higher altitudes. Using gust data generated with a convective boundary layer large eddy simulation, cross-correlation analysis between the time evolution of the frequency content corresponding to elevated wind gusts and the pressure on the ground below were investigated. Several common features of the pressures caused by elevated gusts were identified. These features were used to develop a tracking program that monitors fast moving high amplitude pressure fluctuations and to design a ground based pressure sensing array. The array design and tracking software was used to identify several new gust events within the simulated atmosphere.

3:15

4pPA9. Response of a channel in a semi-infinite stratified medium. Ambika Bhatta, Hui Zhou, Nita Nagdewate, Charles Thompson, and Kavitha Chandra (ECE, UMass, 1 University Ave., Lowell, MA 01854, ambika_bhatta@student.uml.edu)

The presented work focuses in the exact response of two globally reacting surfaces separating a semi-infinite channel from two mediums to a point source when the speed of sound of the host medium is greater than that of the other two mediums. Analytical and numerical image based response will also be discussed in detail for different medium profiles. The modal solution of the 2-D semi-infinite channel of the stratified mediums will be obtained. The Green's function evaluated from the image based reflection coefficient will numerically be compared with the modal solution. The solution approach will be extended for three-dimensional channel. The 3-D response will be discussed in relation with the case of locally reacting surfaces of the channel.

3:30

4pPA10. Spatial coherence function for a wideband acoustic signal. Jericho Cain, Sandra Collier (US Army Res. Lab., 2800 Powder Mill Rd., RDRL-CIE-S, Adelphi, MD 20783, jericho.cain.ctr@mail.mil), Vladimir Ostashev, and D. Keith Wilson (U.S Army Engineer Res. and Development Ctr., Hanover, NH)

Atmospheric turbulence has a significant impact on acoustic propagation. It is necessary to account for this impact in order to study noise propagation, sound localization, and for the development of new remote sensing methods. A solution to a set of recently derived closed form equations for the spatial coherence function of a broadband acoustic pulse propagating in a turbulent atmosphere without refraction and with spatial fluctuations in the wind and temperature fields is presented. Typical regimes of the atmospheric boundary layer are explored.

3:45

4pPA11. Acoustic propagation over a complex site: A parametric study using a time-domain approach. Didier Dragna and Philippe Blanc-Benon (LMFA, Ecole Centrale de Lyon, 36 Ave. Guy de Collongue, Ecully, France, philippe.blanc-benon@ec-lyon.fr)

The influence of the ground characteristics and the meteorological conditions on the acoustic propagation of impulse signals above a complex site is studied. For that, numerical simulations using a finite-difference time-domain solver in curvilinear coordinates [Dragna *et al.*, JASA **133**(6), 3751–3763 (2013)] are performed. The reference site is a railway site in la Veuve near Reims, France, with a non-flat terrain and a mixed impedance ground, where outdoor measurements were performed in May 2010. Comparisons between the experimental data and the numerical results will be reported both in frequency domain and time domain. First, it will be shown that the numerical predictions are in a good agreement with the measured energy spectral densities and waveforms of the acoustic pressure. Second, the impacts of the variations of the ground surface impedances, of the topography and the wind direction will be analyzed.

4:00

4pPA12. The high-order parabolic equation to solve propagation problems in aeroacoustics. Patrice Malbéqui (CFD and aeroAcoust., ONERA, 29, Ave. de la Div. Leclerc, Châtillon 92350, France, patrice.malbequi@onera.fr)

The parabolic equation (PE) has proved its capability to deal with the long range sound propagation in the atmosphere. It also represents an attractive alternative to the ray model to handle duct propagation in high frequencies, for the noise radiated by the nacelle of aero-engines. It was recently shown that the High-Order Parabolic Equation (HOPE), based on a Padé expansion with an order of 5, significantly increases the aperture angle of propagation compared to the standard and the Wide-Angle PEs, allowing prediction close to cut-off frequency of the duct. This paper concerns the propagation using the HOPE in heterogeneous flows, including boundary layers above a wall and in shear layers. The thickness of the boundary layer is about dozens of centimeters while outside it, the Mach number reaches 0.5. The boundary layer effects are investigated showing the refraction effects on a range propagation of 30 m, up to 4 kHz. In the shear layer, discontinuities in the directivity patterns occur significant differences of the directivity patterns occur. Comparisons with the Euler solutions are considered, establishing the domain of application of the HOPE on a set of flow configurations, including beyond its theoretical limits. [Work supported by Airbus-France.]

4:15

4pPA13. Noise and flow measurement of serrated cascade. Kunbo Xu and Qiao Weiyang (School of Power and Energy, Northwestern Polytechnical Univ., No.127 Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, 364398100@qq.com)

This study concerns the mechanisms of the turbulent broadband noise reduction for cascade with the trailing edge serrations. The turbulence spatio-temporal information were measured with 3D hot-wire and the noise results were acquired with a line array. The experiment is carried out in the Northwestern Polytechnical University low speed open jet wind tunnel. It is showed the spreading rate of the wake and the decay rate of the wake centerline velocity deficit increased with serrated edge compared to the straight edge, shedding vortex peaks appeared in the wake, and the three components of velocity changed differently with serrated trailing edge. Serrated trailing edge structure could reduce the radiated noise was proofed by noise results, and some peaks appeared in downstream of the cascade.

THURSDAY AFTERNOON, 8 MAY 2014 555 A/B, 1:30 P.M. TO 5:00 P.M.

Session 4pPP

Psychological and Physiological Acoustics: Role of Medial Olivicochlear Efferents in Auditory Function

Magdalena Wojtczak, Cochair
Psychology, Univ. of Minnesota, 1237 Imperial Ln., New Brighton, MN 55112

Enrique A. Lopez-Poveda, Cochair
Inst. of Neurosci. of Castilla y Leon, Univ. of Salamanca, Calle Pintor Fernando Gallego 1, Salamanca 37007, Spain

Chair's Introduction—1:30

Invited Papers

1:35

4pPP1. Medial olivocochlear efferent effects on auditory responses. John J. Guinan (Eaton Peabody Lab, Mass. Eye & Ear Infirmary, Harvard Med. School, 243 Charles St., Boston, MA 02114, jjg@epl.meei.harvard.edu)

Medial Olivocochlear (MOC) inhibition in one ear can be elicited by sound in either ear. Curiously, the ratio of ipsilateral/contralateral inhibition depends on sound bandwidth; the ratio is ~2 for narrow-band sounds but ~1 for wide-band sounds. Reflex amplitude also depends on elicitor bandwidth and increases as bandwidth is increased, even when elicitor-sound energy is held constant. After elicitor onset (or offset), nothing changes for 20–30 ms and then MOC inhibition builds up (or decays) over 100–300 ms. MOC inhibition has typically been measured in humans by its effects on otoacoustic emissions (OAEs). Problems in such OAE studies include inadequate signal-to-noise ratios (SNRs) and inadequate separation of MOC effects from middle-ear-muscle effects. MOC inhibition reduces basilar-membrane responses more at low levels than high levels, which increases the response SNRs of higher-level signals relative to lower-level background noises, and reduces noise-induced adaptation. The net effect is expected to be increased intelligibility of sounds such as speech. Numerous studies have looked for such perceptual benefits of MOC activity with mixed results. More work is needed to determine whether the differing results are due to experimental conditions (e.g., the speech and noise levels used) or to methodological weaknesses. [Work supported by NIH-RO1DC005977.]

4p THU. PM

1:55

4pPP2. Shelter from the Glutamate storm: Loss of olivocochlear efferents increases cochlear nerve degeneration during aging.
M. Charles Liberman and Stephane F. Maison (Eaton Peabody Labs., Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston, MA, MA 02114, charles_liberman@meei.harvard.edu)

The olivocochlear (OC) feedback pathways include one population, the medial (M)OC projections to outer hair cells, which forms a sound-evoked inhibitory reflex that can reduce sound-induced cochlear vibrations, and a second population, the lateral (L)OC projections to the synaptic zone underneath the inner hair cells, that can modulate the excitability of the cochlear nerve terminals. Although there is ample evidence of OC-mediated protective effects from both of these systems when the ear is exposed to intense noise, the functional significance of this protection is questionable in a pre-industrial environment where intense noise was not so commonplace. We have re-evaluated the phenomenon of OC-mediated protection in light of recent work showing that acoustic exposure destroys cochlear neurons at sound pressure levels previously considered atraumatic, because they cause no permanent hair cell loss or threshold shift. We have shown that loss of OC innervation at a young age causes the cochlea to age a greatly accelerated rate, even without purposeful noise exposure, when aging is measured by the loss of synaptic connections between cochlear nerve fibers and hair cells. Possible relevance to hearing-in-noise problems of the elderly will be discussed.

2:15

4pPP3. Peripheral effects of the cortico-olivocochlear efferent system.
Paul H. Delano (Otolaryngol. Dept., Universidad de Chile, Independencia 1027, Santiago 8380453, Chile, phdelano@gmail.com), Gonzalo Terreros, and Luis Robles (Physiol. and Biophys., ICBM, Universidad de Chile, Santiago, Chile)

The auditory efferent system comprises descending pathways from the auditory cortex to the cochlea, allowing modulation of sensory processing even at the most peripheral level. Although the presence of descending circuits that connect the cerebral cortex with olivocochlear neurons have been reported in several species, the functional role of the cortico-olivocochlear efferent system remains largely unknown. We have been studying the influence of cortical descending pathways on cochlear responses in chinchillas. Here, we recorded cochlear microphonics and auditory-nerve compound action potentials in response to tones (1–8 kHz; 30–90 dB SPL) before, during, and after auditory-cortex lidocaine or cooling inactivation ($n = 20$). In addition, we recorded cochlear potentials in the presence and absence of contralateral noise, before, during, and after auditory-cortex micro-stimulation (2-50 μA, 32 Hz rate) ($n = 15$). Both types of auditory-cortex inactivation produced changes in the amplitude of cochlear potentials. In addition, in the microstimulation experiments, we found an increase of the suppressive effects of contralateral noise in neural responses to 2–4 kHz tones. In conclusion, we demonstrated that auditory-cortex basal activity exerts tonic influences on the olivocochlear system and that auditory-cortex electrical micro-stimulation enhances the suppressive effects of the acoustic evoked olivocochlear reflex. [Work supported by FONDECYT 1120256; FONDECYT 3130635 and Fundacion Puelma.]

2:35

4pPP4. Does the efferent system aid with selective attention?
Dennis McFadden (Psych., Univ. of Texas, 108 E. Dean Keeton A8000, Austin, TX 78712-1043, mcfadden@psy.utexas.edu), Kyle P. Walsh (Psych., Univ. of Minnesota, Minneapolis, MN), and Edward G. Pasanen (Psych., Univ. of Texas, Austin, TX)

To study whether attention and inattention lead to differential activation of the olivocochlear (OC) efferent system, a cochlear measure of efferent activity was collected while human subjects performed behaviorally under the two conditions. Listeners heard two independent, simultaneous strings of seven digits, one spoken by a male and the other by a female, and at the end of some trials (known in advance), they were required to recognize the middle five digits spoken by the female. Interleaved with the digits were one stimulus that evokes a stimulus-frequency otoacoustic emission (SFOAE) and another that activates the OC system—a 4-kHz tone (60 dB SPL, 300 ms in duration) and a wideband noise (1.0–6.0 kHz, 25 dB spectrum level, 250 ms in duration, beginning 50 ms after tone onset). These interleaved sounds, used with a double-evoked procedure, permitted the collection of a nonlinear measure called the nSFOAE. When selective attention was required behaviorally, the magnitude of the nSFOAE to tone-plus-noise differed by 1.3–4.0 dB compared to inattention. Our interpretation is that the OC efferent system was more active during attention than during relative inattention. Whether or how this efferent activity actually aided behavioral performance under attention is not known.

2:55

4pPP5. Behavioral explorations of cochlear gain reduction.
Elizabeth A. Strickland, Elin Roverud, and Kristina DeRoy Milvae (Speech, Lang., and Hearing Sci., Purdue Univ., 500 Oval Dr., West Lafayette, IN 47907, estrick@purdue.edu)

Physiological measures have shown that the medial olivocochlear reflex (MOCR) decreases the gain of the cochlear active process in response to ipsilateral or contralateral sound. As a first step to determining its role in human hearing in different environments, our lab has used psychoacoustical techniques to look for evidence of the MOCR in behavioral results. Well-known forward masking techniques that are thought to measure frequency selectivity and the input/output function at the level of the cochlea have been modified so that the stimuli (masker and signal) are short enough that they should not evoke the MOCR. With this paradigm, a longer sound (a precursor) can be presented before these stimuli to evoke the MOCR. The amount of threshold shift caused by the precursor depends on its duration and its frequency relative to the signal in a way that supports the hypothesis that the precursor has reduced the gain of the cochlear active process. The magnitude and time course of gain reduction measured across our studies will be discussed. The results support the hypothesis that one role of the MOCR may be to adjust the dynamic range of hearing in noise. [Work supported by NIH(NIDCD)R01 DC008327, T32 DC000030-21, and Purdue Research Foundation.]

3:15–3:30 Break

3:30

4pPP6. Challenges in exploring the role of medial olivocochlear efferents in auditory tasks via otoacoustic emissions. Magdalena Wojtczak (Psych., Univ. of Minnesota, 1237 Imperial Ln., New Brighton, MN 55112, wojtc001@umn.edu), Jordan A. Beim, and Andrew J. Oxenham (Psych., Univ. of Minnesota, Minneapolis, MN)

A number of recent psychophysical studies have hypothesized that the activation of medial olivocochlear (MOC) efferents plays a significant role in forward masking. These hypotheses are based on general similarities between spectral and temporal characteristics exhibited by some psychophysical forward-masking results and by effects of efferent activation measured using physiological methods. In humans, noninvasive physiological measurements of otoacoustic emissions have been used to probe changes in cochlear responses due to MOC efferent activation. The aim of this study was to verify our earlier efferent-based hypothesis regarding the dependence of psychophysical forward masking of a 6-kHz probe on the phase curvature of harmonic-complex maskers. The ear-canal pressure for a continuous 6-kHz probe was measured in the presence and absence of Schroeder-phase complexes used as forward maskers in our previous psychophysical study. Changes in the ear-canal pressure were analyzed using methods for estimating the effects of efferent activation on stimulus frequency otoacoustic emissions under the assumption that changes in cochlear gain due to efferent activation will be reflected in changes in the magnitude and phase of the emission. Limitations and challenges in relating effects of feedback-based reflexes to psychophysical effects will be discussed. [Work supported by NIH grant R01DC010374.]

3:50

4pPP7. The function of the basilar membrane and medial olivocochlear (MOC) reflex mimicked in a hearing aid algorithm. Tim JÃrgens (Dept. of Medical Phys. and Acoust., Cluster of Excellence Hearing4all, UniversitÃt Oldenburg, Carl-von-Ossietzky Str. 9-11, Oldenburg 26121, Germany, tim.juergens@uni-oldenburg.de), Nicholas R. Clark, Wendy Lecluyse (Dept. of Psych., Univ. of Essex, Colchester, United Kingdom), and Meddis Ray (Dept. of Psych., Univ. of Essex, Colchester, Germany)

The hearing aid algorithm “BioAid” mimics two basic principles of normal hearing: the instantaneous compression of the basilar membrane and the efferent feedback of the medial olivocochlear (MOC) reflex. The design of this algorithm aims at restoring those parts of the auditory system, which are hypothesized to dysfunction in the individual listener. In the initial stage of this study individual computer models of three hearing-impaired listeners were constructed. These computer models reproduce the listeners’ performance in psychoacoustic measures of (1) absolute thresholds, (2) compression, and (3) frequency selectivity. Subsequently, these computer models were used as “artificial listeners.” Using BioAid as a front-end to the models, parameters of the algorithm were individually adjusted with the aim to ‘normalize’ the model performance on these psychoacoustic measures. In the final stage of the study, the optimized hearing aid fittings were evaluated with the three hearing-impaired listeners. The aided listeners showed the same qualitative characteristics of the psychoacoustic measures as the aided computer models: near-normal absolute thresholds, steeper compression estimates and sharper frequency selectivity curves. A systematic investigation of the effect of compression and the MOC feedback in the algorithm revealed that both are necessary to restore performance. [Work supported by DFG.]

4:10

4pPP8. Mimicking the unmasking effects of the medial olivo-cochlear efferent reflex with cochlear implants. Enrique A. Lopez-Poveda and Almudena Eustaquio-Martin (Inst. of Neurosci. of Castilla y Leon, Univ. of Salamanca, Calle Pintor Fernando Gallego 1, Salamanca, Salamanca 37007, Spain, ealopezpoveda@usal.es)

In healthy ears, cochlear sensitivity and tuning are not fixed; they vary depending on the state of activation of medial olivo-cochlear (MOC) efferent fibers, which act upon outer hair cells modulating the gain of the cochlear amplifier. MOC efferents may be activated in a reflexive manner by ipsilateral and contralateral sounds. Activation of the MOC reflex (MOCR) is thought to unmask sounds by reducing the adaptation of auditory nerve afferent fibers response to noise. This effect almost certainly improves speech recognition in noise. Furthermore, there is evidence that contralateral stimulation can improve the detection of pure tones embedded in noise as well as speech intelligibility in noise probably by activation of the contralateral MOCR. The unmasking effects of the MOCR are unavailable to current cochlear implant (CI) users and this might explain part of their difficulty at understanding speech in noise compared to normal hearing subjects. Here, we present preliminary results of a bilateral CI sound-coding strategy that mimics the unmasking benefits of the ipsilateral and contralateral MOCR. [Work supported by the Spanish MINECO and MED-EL GmbH.]

Contributed Papers

4:30

4pPP9. Mice with chronic medial olivocochlear dysfunction do not perform as predicted by common hypotheses about the role of efferent cochlear feedback in hearing. Amanda Lauer (Otolaryngology-HNS, Johns Hopkins Univ. School of Medicine, 515 Traylor, 720 Rutland Ave., Baltimore, MD 21205, alauer2@jhmi.edu)

Mice missing the alpha9 nicotinic acetylcholine receptor subunit (A9KO) show a lack of classic efferent effects on cochlear activity; however, behavioral and physiological studies in these mice have failed to support common hypotheses about the role of efferent feedback in auditory function. A9KO mice do not show deficits detecting or discriminating tones in noise. These mice also do not appear to be more susceptible to age-related hearing loss, and they do not show increased auditory brainstem response thresholds when chronically exposed to moderate-level noise. A9KO mice do show increased susceptibility to temporal processing deficits, especially when exposed to environmental noise. Furthermore, A9KO mice show extremely variable, and sometimes poor, performance when discriminating changes in the location of broadband sounds in the horizontal plane. Temporal and spatial processing deficits may be attributable to abnormal or poorly optimized representation of acoustic cues in the central auditory pathways. These results are consistent with experiments in humans that suggest artificial stimulation of medial olivocochlear efferents overestimates the actual activation of these pathways. Thus, the primary role of medial olivocochlear efferent feedback may be to regulate input from the cochlea to the brain (and within the brain) to maintain an optimal, calibrated representation of sounds.

4:45

4pPP10. Time-course of recovery from the effects of a notched-noise on the ear-canal pressure at different frequencies. Kyle P. Walsh and Magdalena Wojtczak (Psych., Univ. of Minnesota, 75 East River Rd., Minneapolis, MN 55455, kpwalsh@umn.edu)

Different methods for estimating the effect of the medial olivocochlear reflex (MOCR) on stimulus-frequency otoacoustic emissions (SFOAEs) in humans appear to yield different estimates of the time-course of recovery from the effect. However, it is uncertain whether the observed differences in recovery times were due to differences in the methods used to extract the changes in SFOAEs, due to the fact that different feedback-based reflexes—MOCR or the middle ear muscle reflex (MEMR)—were activated, or due to the dependence of recovery from the activated reflex on the probe frequency. In this study, the ear-canal pressure was measured for continuous probes with frequencies of 1, 2, 4, and 6 kHz, in the presence and absence of an ipsilateral notched-noise elicitor. Changes in the magnitude and phase of the ear-canal pressure were extracted to estimate recovery times from the effects of the elicitor. The results showed that the recovery time increased with increasing probe frequency—from about 380 ms at 1 kHz to about 1500 ms at 6 kHz, on average. The measurements also were repeated for each of the probe frequencies paired with a simultaneous 500-Hz tone to examine the role of the MEMR. [Work supported by NIH grant R01DC010374.]

THURSDAY AFTERNOON, 8 MAY 2014 — 553 A/B, 1:30 P.M. TO 4:55 P.M.

Session 4pSA

Structural Acoustics and Vibration and Physical Acoustics: Acoustics of Cylindrical Shells II

Sabih I. Hayek, Cochair
Eng. Sci., Penn State, 953 McCormick Ave., State College, PA 16801-6530

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

Invited Papers

1:30

4pSA1. A study of multi-element/multi-path concentric shell structures to reduce noise and vibration. Donald B. Bliss, David Raudales, and Linda P. Franzoni (Dept. of Mech. Eng. and Mater. Sci., Duke Univ., Durham, NC 27708, dbb@duke.edu)

Vibration transmission and noise can be reduced by dividing a structural barrier into several constituent subsystems with separate, elastically coupled, wave transmission paths. Multi-element/multi-path (MEMP) structures utilize the inherent dynamics of the system, rather than damping, to achieve substantial wide-band reduction in the low frequency range, while satisfying constraints on static strength and weight. The increased complexity of MEMP structures provides a wealth of opportunities for reduction, but the approach requires rethinking the structural design process. Prior analytical and experimental work, reviewed briefly, focused on simple beam systems. The current work extends the method to elastically coupled concentric shells, and is the first multi-dimensional study of the concept. Subsystems are modeled using a modal decomposition of the thin shell equations. Axially discrete azimuthally continuous elastic connections occur at regular intervals along the concentric shells. Simulations show the existence of robust solutions that provide large wide-band reductions. Vibratory force and sound attenuation are achieved through several processes acting in concert: different subsystem wave speeds, mixed boundary conditions at end points, interaction through elastic couplings, and stop band behavior. The results show the concept may have application in automotive and aerospace vehicles, and low vibration environments such as sensor mounts.

1:50

4pSA2. Scattering from a cylindrical shell with an internal mass. Andrew Norris and Alexey S. Titovich (Mech. and Aerosp. Eng., Rutgers Univ., 98 Brett Rd., Piscataway, NJ 08854, norris@rutgers.edu)

Perhaps the simplest approach to modeling acoustic scattering from objects with internal substructure is to consider a cylindrical shell with an internal mass attached by springs. The earliest analyses, published in JASA in 1992, by Achenbach *et al.* and by Guo assumed one and two springs, respectively. Subsequent studies examined the effects of internal plates and more sophisticated models of substructure. In this talk we reconsider the Achenbach—Guo model but for an arbitrary number, say J, of axisymmetrically distributed stiffeners. The presence of a springs-mass substructure breaks the cylindrical symmetry, coupling all azimuthal modes. Our main result provides a surprisingly simple form for the scattering solution for time harmonic incidence. We show that the scattering, or T-matrix, decouples into the sum of the T-matrix for the bare shell plus J matrices each defined by an infinite vector. In addition, an approximate expression is derived for the frequencies of the quasi-flexural resonances induced by the discontinuities on the shell, which excite subsonic shell flexural waves. Some applications of the model to shells with specified long wavelength effective bulk modulus and density will be discussed. [Work supported by ONR.]

2:10

4pSA3. Active noise control for cylindrical shells using a sum of weighted spatial gradients (WSSG) control metric. Pegah Aslani, Scott D. Sommerfeldt, Yin Cao (Dept. of Phys. and Astronomy, N203 ESC Brigham Young Univ., Provo, UT 84602-4673, pegah.aslani@gmail.com), and Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT)

There are a number of applications involving cylindrical shells where it is desired to attenuate the acoustic power radiated from the shell, such as from an aircraft fuselage or a submarine. In this paper, a new active control approach is outlined for reducing radiated sound power from structures using a weighted sum of spatial gradients (WSSG) control metric. The structural response field associated with the WSSG has been shown to be relatively uniform over the surface of both plates and cylindrical shells, which makes the control method relatively insensitive to error sensor location. It has also been shown that minimizing WSSG is closely related to minimizing the radiated sound power. This results in global control being achieved using a local control approach. This paper will outline these properties of the WSSG control approach and present control results for a simply supported cylindrical shell showing the attenuation of radiated sound power that can be achieved.

2:30

4pSA4. Causality and scattering from cylindrical shells. James G. McDaniel (Mech. Eng., Boston Univ., 110 Cummington St., Boston, MA 02215, jgm@bu.edu)

Acoustic scattering from a cylindrical shell is required to be causal, so that the incident wave must precede the scattered wave that it creates. In the frequency domain, this statement may be explored by forming a frequency-dependent complex-valued reflection coefficient that relates the scattered wave to the incident wave. The real and imaginary parts of the reflection coefficient must therefore satisfy Hilbert Transform relations that involve integrals over frequency. As a result, one may find the real part of the reflection coefficient given only its imaginary part over a frequency range, and vice-versa. The reflection coefficient is not required to be minimum phase and rarely is minimum phase, so the causality condition cannot be used directly to estimate the phase of the reflection coefficient from its magnitude. However, the effective impedance associated with the reflection coefficient is required to be minimum phase. An approach is presented for using these relations to estimate the phase of a reflection coefficient given only its magnitude. Examples are presented that illustrate these relationships for cylindrical shells.

2:50

4pSA5. Frequency domain comparisons of different analytical and computational radiated noise solutions for point-excited cylindrical shells. 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)

Among a multitude of diverse applications, the acoustics of cylindrical shells is also an important area of study for its applicability to and representation of many US Navy undersea vehicles and systems. Examination of structural acoustic predictions of cylindrical-shell-based system designs are frequently made using a variety of analytical and computational approaches including closed-form 3D elasticity, numerous kinematic plate/shell theories, Finite Element Analysis (FEA), Energy-based FEA (EFEA) coupled with Energy Boundary Element Analysis (EBEA), and Statistical Energy Analysis (SEA). Each of these approaches has its own set of assumptions, advantages, and applicable frequency range which can make for confusion. This paper presents radiated noise solutions in the area of cylindrical shell structural acoustics from the above list of methodologies for the canonical problem of a point-excited, finite cylindrical shell with/without fluid loading. Specifically, far-field radiated sound power predictions for cylindrical shells using many different classical analytical and modern day numerical approaches (i.e., 3D elasticity, closed form plate and shell theory solutions FEA, EFEA/EBEA, SEA) are made and compared. Of particular interest for this comparison is the applicable frequency regimes for each solution and also how the solution approaches compare/transition from one to the other over a wide frequency range.

3:10–3:30 Break

3:30

4pSA6. Applications of interior fluid-loaded orthotropic shell theory for noise control and cochlear mechanics. Karl Grosh, Suyi Li, and Kon-Well Wang (Dept. of Mech. Eng., Univ. of Michigan, 2350 Hayward St., Ann Arbor, MI 48109-2125, grosh@umich.edu)

The vibration of shells with heavy interior fluid loading is a classical theory, as analyzed nearly 30 years ago by Fuller and Fahy in a series of seminal papers. Wave propagation for interiorly filled hydraulic lines, biological blood vessels, and pipelines represent classes of well-studied problems. In this paper we consider the application of this theory to two specific and seemingly disparate problems. The theory for interiorly fluid-loaded finite orthotropic shells with heavy interior fluid loading subject to end loading and with stiff end-cap terminations will be presented and compared to detailed experimental results. Application of this theory to the development of transfer matrices for developing networks of interconnected units of these systems (including the possibility of fluid flow between vessels) will be presented along with a discussion of the effects of fluid compressibility for the mechanics of outer hair cells of the mammalian cochlea.

3:50

4pSA7. Acoustic scattering from finite bilaminar cylindrical shells-directivity functions. Sabih I. Hayek (Eng. Sci., Penn State, 953 McCormick Ave., State College, PA 16801-6530, sihesm@engr.psu.edu) and Jeffrey E. Boisvert (NAVSEA Div. Newport, NUWC, Newport, RI)

The spectra of the acoustic scattered field from a normally insonified finite bilaminar cylindrical shell has been previously analyzed using the exact theory of three-dimensional elasticity (J. Acoust. Soc. Am. **134**, 4013 (2013)). The two shell laminates, having the same lateral dimensions but different radii and material properties, are perfectly bonded. The finite bilaminar shell is submerged in an infinite

acoustic medium and is terminated by two semi-infinite rigid cylindrical baffles. The shell has shear-diaphragm supports at the ends $z = 0$, L and is internally filled with another acoustic medium. The bilaminar shell is insonified by an incident plane wave at an oblique incidence angle. The scattered acoustic farfield directivity function is evaluated for various incident wave frequencies and for a range of shell thicknesses, lengths, radii, and material properties. A uniform steel and a bilaminar shell made up of an outer elastomeric material bonded to an inner steel shell are analyzed to study the influence of elastomeric properties on the directivity functions. [Work supported by NAVSEA Division Newport under ONR Summer Faculty Program.]

Contributed Papers

4:10

4pSA8. Coupled vibrations in hollow cylindrical shells of arbitrary aspect ratio. Boris Aronov (BTech Acoust. LLC, Fall River, MA) and David A. Brown (Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723, dbAcoustics@cox.net)

Vibrations of hollow cylinders have been the subject of considerable interest for many years. Piezoelectric cylinders offer a convenient system to study the vibration mode shapes, resonance frequencies and their mode coupling do to the ability to strongly and symmetrically excite extensional circumferential and axial vibration modes as well as flexural bending axial modes. While the mode repulsion of coupled circumferential and axial modes is generally widely known, their interaction gives rise to tubular flexural resonances in cylinders of finite thickness. Junger *et al*. [JASA **26**, 709–713 (1954)] appears to have been first to discredit the notion of a forbidden zone, a frequency band free of resonant modes, as being an artifact of treating thin cylinders in the membrane limit. Aronov [JASA **125**(2), 803–818 (2009)] showed experimental and theoretical proof of the presence of resonant modes throughout the spectrum as a result of the extensional mode coupling induced symmetric tubular bending modes in cylinders and their relationships as a function of different piezoelectric polarizations. That analysis used the energy method and the Euler-Lagrange equations based on the coupling of assumed modes of vibration and the synthesis of results using equivalent electromechanical circuits. This paper aims to both summarize and generalize those results for the applicability of passive cylindrical shells.

4:25

4pSA9. Attenuation of noise from impact pile driving in water using an acoustic shield. Per G. Reinhall, Peter H. Dahl, and John T. Dardis (Mech. Eng., Univ. of Washington, Stevens Way, Box 352600, Seattle, WA 98195, tdardis@u.washington.edu)

Offshore impact pile driving produces extremely high sound levels in water. Peak acoustic pressures from the pile driving operation of ~10^3 Pa at a range of 3000 m, ~10^4 Pa at a range of 60 m, and ~10^5 Pa at a range of 10 m have been measured. Pressures of these magnitudes can have negative effects on both fish and marine mammals. Previously, it was shown that the primary source of sound originates from radial expansion of the pile as a compression wave propagates down the pile after each strike. As the compression wave travels it produces an acoustic field in the shape of an axisymmetric cone, or Mach cone. The field associated with this cone clearly dominates the peak pressures. In this paper, we present an evaluation of the effectiveness of attenuating pile driving noise using an acoustic shield. In order to fully evaluate the acoustic shield, we provide results from finite element modeling and simple plane wave analysis of impact pile driving events with and without a noise shield. This effort is supported by the findings from a full-scale pile driving experiment designed to evaluate the effectiveness of the noise shield. Finally, we will discuss methods for improving the effectiveness of the acoustic shield.

4:40

4pSA10. Free and forced vibrations of hollow elastic cylinders of finite length. D. D. Ebenezer, K. Ravichandran (Naval Physical and Oceanogr. Lab, Thrikkakara, Kochi, Kerala 682021, India, d.d.ebenezer@gmail.com), and Chandramouli Padmanabhan (Indian Inst. of Technol., Madras, Chennai, Tamil Nadu, India)

An analytical model of axisymmetric vibrations of hollow elastic circular cylinders with arbitrary boundary conditions is presented. Free vibrations of cylinders with free or fixed boundaries and forced vibrations of cylinders with specified non-uniform displacement or stress on the boundaries are considered. Three series solutions are used and each term in each series is an exact solution to the exact governing equations of motion. The terms in the expressions for components of displacement and stress are products of Bessel and sinusoidal functions and are orthogonal to each other. Complete sets of functions in the radial and axial directions are formed by terms in the first series and the other two, respectively. It is therefore possible to satisfy arbitrary boundary conditions. It is shown that two terms in each series are sufficient to determine several resonance frequencies of cylinders with certain specified boundary conditions. The error is less than 1% for free cylinders. Numerical results are also presented for forced vibration of hollow steel cylinders of length 10 mm and outer diameter 10 mm with specified normal displacement or stress. Excellent agreement with finite element results is obtained at all frequencies up to 1 MHz. Convergence of the series is also discussed.

Session 4pSC

Speech Communication: Special Populations and Clinical Considerations

Sarah H. Ferguson, Chair
Commun. Sci. and Disorders, Univ. of Utah, 390 South 1530 East, Rm. 1201, Salt Lake City, UT 84112

Contributed Papers

1:30

4pSC1. Internal three dimensional tongue motion during "s" and "sh" from tagged magnetic resonance imaging; control and glossectomy motion. Joseph K. Ziemba, Maureen Stone, Andrew D. Pedersen, Jonghye Woo (Neural and Pain Sci., Univ. of Maryland Dental School, 650 W. Baltimore St., Rm. 8207, Orthodontics, Baltimore, MD 21201, mstone@umaryland.edu), Fangxu Xing, and Jerry L. Prince (Elec. and Comput. Eng., Johns Hopkins Univ., Baltimore, MD)

This study aims to ascertain the effects of tongue cancer surgery (glossectomy) on tongue motion during the speech sounds "s" and "sh." Subjects were one control and three glossectomies. The first patient had surgery closed with sutures. The second had sutures plus radiation, which produces fibrosis and stiffness. The third was closed with an external free flap, and is of particular interest since he has no direct motor control of the flap. Cine and tagged-MRI data were recorded in axial, coronal and sagittal orientations at 26 fps. 3D tissue point motion was tracked at every time-frame in the word. 3D displacement fields were calculated at each time-frame to show tissue motion during speech. A previous pilot study showed differences in "s" production [Pedersen *et al.*, JASA (2013)]. Specifically, subjects differed in internal tongue motion pattern, and the flap patient had unusual genioglossus lengthening patterns. The "s" requires a midline tongue groove, which is challenging for the patients. This study continues that effort by adding the motion of "sh," because "sh" does not require a midline groove and may be easier for the patients to pronounce. We also add more muscles, to determine how they interact to produce successful motion. [This study was supported by NIH R01CA133015.]

1:45

4pSC2. An acoustic threshold for third formant in American English /r/. Sarah M. Hamilton, Suzanne E. Boyce, Leah Scholl, and Kelsey Douglas (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Mail Location 379Cincinnati, OH 45267, Suzanne.Boyce@uc.edu)

It is well known that a low F3 is the most salient acoustic feature of American English /r/, and that the degree of F3 lowering is correlated with the degree to which /r/ is perceptually acceptable to native listeners as a "good" vs. "misarticulated" /r/. Identifying the point at which F3 lowering produces a "good" /r/ would be helpful in remediation of /r/-production difficulties in children and second language learners. Such a measure would require normalization across speakers. Hagiwara (1995) observed that F3 for /r/ in competent adult speakers was at or below 80% of the average vowel frequencies for a given speaker. In this study, we investigate whether children's productions start to sound "good" when they lower F3 to the 80% demarcation level or below. Words with /r/ and vowel targets from 20 children with a history of /r/ misarticulation were extracted from acoustic records of speech therapy sessions. Three experienced clinicians judged correctness of /r/ productions. Measured F3's at the midpoint of /r/ and a range of vowels were compared for these productions. Preliminary findings suggest that the 80% level is a viable demarcation point for good vs. misarticulated articulation of /r/.

2:00

4pSC3. Prosodic variability in the speech of children who stutter. Timothy Arbisi-Kelm, Julia Hollister, Patricia Zebrowski, and Julia Gupta (Commun. Sci. and Disord., Univ. of Iowa, Wendell Johnson Speech and Hearing Ctr., Iowa City, IA 52242, timothy-arbisi-kelm@uiowa.edu)

Developmental stuttering is a heterogeneous language disorder characterized by persistent speech disruptions, which are generally realized as repetitions, blocks, or prolongations of sounds and syllables (DSM-IV-R, 1994). While previous studies have uncovered ample evidence of deficits in both "higher-level" linguistic planning and "lower-level" motor plan assembly, identifying the relative contribution of the specific factors underlying these deficits has proved difficult. Phrasal prosody represents a point of intersection between linguistic and motoric planning, and therefore a promising direction for stuttering research. In the present study, 12 children who stutter (CWS) and 12 age-matched controls (CWNS) produced sentences varying in length and syntactic complexity. Quantitative measures (F0, duration, and intensity) were calculated for each word, juncture, and utterance. Overall, CWS produced a narrower F0 range across utterance types than did CWNS, while utterance duration did not differ significantly between groups. Within utterances, CWS (but not CWNS) produced a greater degree of preboundary lengthening preceding relative clauses in syntactically complex sentences, as well as higher F0 variability at these juncture points. Such differences suggest that for CWS utterance planning is sensitive to syntactic complexity, possibly reflecting either a deficit in syntactic processing or the relative effects of syntactic processing on a strained processing system.

2:15

4pSC4. Tongue shape complexity for liquids in Parkinsonian speech. Doug H. Whalen (Haskins Labs., 300 George St. Ste. 900, New Haven, CT 06511, whalen@haskins.yale.edu), Katherine M. Dawson, Micalle Carl (Speech-Language-Hearing Sci., City Univ. of New York, New York, NY), and Khalil Iskarous (Dept. of Linguist, Univ. of Southern California, Los Angeles, CA)

Parkinson's disease (PD) is a neurological disorder characterized by the degeneration of dopaminergic neurons. Speech impairments in PD are characterized by slowed muscle activation, muscle rigidity, variable rate, and imprecise consonant articulation. Complex muscular synergies are necessary to coordinate tongue motion for linguistic purposes. Our previous work showed that people with PD had an altered rate of change in tongue shape during vowel to consonant transitions, but differences were small, perhaps due to the simplicity of the speech task. In order to test sentences, four PD participants and three older controls were imaged using ultrasound. They repeated sentences from the Rainbow Passage. Tongue shape complexity in liquids and adjacent vowels was assessed by their bending energy [Young *et al.*, Info. Control **25**(4), 357–370 (1974)]. Preliminary results show that bending energy was higher in liquids than in vowels, and higher in controls than PD speakers. Production of liquids typical requires a flexible tongue shape; these PD speakers show reduced flexibility that is nonetheless compensated sufficiently for the production of intelligible speech. Implications for speech motor control and for PD evaluation will be discussed.

2:30

4pSC5. VocaliD: Personal voices for augmented communicators. H Timothy Bunnell (Ctr. for Pediatric Auditory and Speech Sci., Alfred I. duPont Hospital for Children, 1701 Rockland Rd., Wilmington, DE 19807, bunnell@asel.udel.edu) and Rupal Patel (Dept. of Speech Lang. Pathol. and Audiol., Northeastern Univ., Boston, MA)

The goal of the VocaliD project (for vocal identity) is to develop personalized synthetic voices for children and adults who rely on speech generating devices (SGDs) for verbal communication. Our approach extracts acoustic properties related to source, vocal tract, or both from a target talker's disordered speech (whatever sounds they can still produce) and applies these features to a synthetic voice that was created from a surrogate voice donor who (ideally) is similar in age, size, gender, etc. The result is a synthetic voice that contains as much of the vocal identity of the target talker as possible yet the speech clarity of the surrogate talker's synthetic voice. To date, we have deployed several synthetic voices using this technology. Three case studies will be presented to illustrate the methods used in voice generation and the results from three pediatric SGD users. We will also describe plans to greatly extend our database of surrogate voice donor speech, allowing us to better match regional/dialectical features to the needs of the target SGD users.

2:45

4pSC6. Perceptual learning in the laboratory versus real-world conversational interaction. Elizabeth D. Casserly (Dept. of Psych., Trinity College, 300 Summit St., Hartford, CT 06106, elizabeth.casserly@trincoll.edu) and David B. Pisoni (Dept. of Psychol. & Brain Sci., Indiana Univ., Bloomington, IN)

Understanding perceptual learning effects under novel acoustic circumstances, e.g., situations of hearing loss or cochlear implantation, constitutes a critical goal for research in the hearing sciences and for basic perceptual research surrounding spoken language use. These effects have primarily been studied in traditional laboratory settings using stationary subjects, prerecorded materials, and a restricted set of potential subject responses. In the present series of experiments, we extended this paradigm to investigate perceptual learning in a situated, interactive, real-world context for spoken language use. Experiments 1 and 2 compared the learning achieved by normal-hearing subjects experiencing real-time cochlear implant acoustic simulation in either conversation or traditional feedback-based computer training. In experiment 1, we found that interactive conversational subjects achieved perceptual learning equal to that of laboratory-trained subjects for speech recognition in the quiet, but neither group generalized this learning to other domains. Experiment 2 replicated the learning findings for speech recognition in quiet and further demonstrated that subjects given active perceptual exposure were able to transfer their perceptual learning to a novel task, gaining significantly more benefit from the availability of semantic context in an isolated word recognition task than subjects who completed conventional laboratory-based training.

3:00

4pSC7. Spectrotemporal alterations and syllable stereotypy in the vocalizations of mouse genetic models of speech-language disorders. Gregg A. Castellucci (Linguist, Yale Univ., 333 Cedar St., Rm. I-407, New Haven, CT 06511, gregg.castellucci@yale.edu), Matthew J. McGinley, and David A. McCormick (Neurobiology, Yale School of Medicine, New Haven, CT)

Specific language impairment (SLI) and developmental dyslexia (DD) are common speech-language disorders exhibiting a range of phonological and speech motor deficits. Recently, mouse genetic models of SLI (Foxp2) and DD (Dcdc2) have been developed and promise to be powerful tools in understanding the biological basis of these diseases. Surprisingly, no studies of the adult vocalizations—which exhibit the most elaborate and complex call structure—have been performed in these mouse strains. Here, we analyze the male ultrasonic courtship song of Dcdc2 knockout mice and Foxp2 heterozygous knockout mice and compare it to the song of their C57BL/6J background littermates. Preliminary analysis indicates considerable difference between the three groups. For example, Foxp2 heterozygous knockout song contains less frequency modulation and has a reduced syllable inventory in comparison to that of wildtype littermates. The call production and phonological deficits exhibited by these mouse models are reminiscent of the symptoms observed in humans with these disorders.

3:15

4pSC8. Listening effort in bilateral cochlear implants and bimodal hearing. Matthew Fitzgerald, Katelyn Glassman (Otolaryngol., New York Univ. School of Medicine, 550 1st Ave., NBV-5E5, New York, NY 10016, fitz.mb@gmail.com), Sapna Mehta (City Univ. of New York, New York, NY), Keena Seward, and Arlene Neuman (Otolaryngol., New York Univ. School of Medicine, New York, NY)

Many users of bilateral cochlear implants, or of bimodal hearing, report, reduced listening effort when both devices are active relative to a single device. To quantify listening effort in these individuals, we used a dual-task paradigm. In such paradigms, the participant divides attention between a primary and secondary task. As the primary task becomes more difficult, fewer cognitive resources are available for the secondary task, resulting in poorer performance. The primary task was to repeat AzBio sentences in quiet, and in noise. The secondary task was to recall a digit string presented visually before a set of two sentences. As a control, both the primary and secondary tasks were tested alone in a single-task paradigm. Participants were tested unilaterally and bilaterally / bimodally. Relative to the single-task control, scores obtained in the dual-task paradigm were not affected in the primary sentence-recognition task, but were lower on the secondary digit-recall task. This suggests that a dual-task paradigm has potential to quantify listening effort. Some listeners who showed bilateral benefits to speech understanding had higher bilateral than unilateral digit-recall scores. However, there was considerable variability on the digit-recall task, which hinders our ability to draw clear conclusions.

3:30–3:45 Break

3:45

4pSC9. Measurement of spectral resolution and listening effort in people with cochlear implants. Matthew Winn (Dept. of Surgery, Univ. of Wisconsin-Madison, 1500 Highland Ave., Rm. 565, Madison, WI 53705, mwinn83@gmail.com), Ruth Y. Litovsky, and Jan R. Edwards (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, Madison, WI)

Cochlear implants (CIs) provide notably poor spectral resolution, which poses significant challenges for speech understanding, and places greater demands on listening effort. We evaluated a CI stimulation strategy designed to improve spectral resolution by measuring its impact on listening effort (as quantified by pupil dilation, which is considered to be a reliable index of cognitive load). Specifically, we investigated dichotic interleaved processing channels (where odd channels are active in one ear, and even channels are active in the contralateral ear). We used a sentence listening and repetition task where listeners alternated between their everyday clinical CI configurations and the interleaved channel strategy, to test which offered better resolution and demanded less effort. Methods and analyses stemmed from previous experiments confirming that spectral resolution has a systematic impact on listening effort in individuals with normal hearing. Pupil dilation measures were generally consistent with speech perception ($r^2 = 0.48$, $p < 0.001$), suggesting that spectral resolution plays an important role in listening effort for listeners with CIs. When using interleaved channels, both speech perception performance and pupillary responses were variable across individuals, underscoring the need for individualized measurement for CI listeners rather than group analysis, in the pursuit of better clinical fitting.

4:00

4pSC10. Automatic speech recognition of naturalistic recordings in families with children who are hard of hearing. Mark VanDam (Speech & Hearing Sci., Washington State Univ., PO BOX 1495, Spokane, WA 99202, mark.vandam@wsu.edu) and Noah H. Silbert (Commun. Sci. & Disord., Univ. Cincinnati, Cincinnati, OH)

Performance of an automatic speech recognition (ASR) system [LENA Research Foundation, Boulder, CO] has been reported for naturalistic, whole day recordings collected in families with typically developing (TD) children. This report examines ASR performance of the LENA system in

families with children who are hard-of-hearing (HH). Machine-labeled segments were compared with human judges' assessment of talker identity (*child*, *mother*, or *father*), and recordings from families with TD children were compared with families with HH children. Classification models were fit to several acoustic variables to assess decision process differences between machine and human labels and between TD and HH groups. Accuracy and error of both machine and human performance is reported. Results may be useful to improve implementation and interpretation of ASR techniques in terms of special populations such as children with hearing loss. Findings also have implications for very large database applications of unsupervised ASR, especially its application to naturalistic acoustic data.

4:15

4pSC11. Assessing functional auditory performance in hearing-impaired listeners with an updated version of the Modified Rhyme Test.

Douglas Brungart, Matthew J. Makashay, Van Summers, Benjamin M. Sheffield, and Thomas A. Heil (Audiol. and Speech Pathol. Ctr., Walter Reed NMMC, 8901 Wisconsin Ave., Bethesda, MD 20889, douglas.brungart@us.army.mil)

Pure-tone audiometric thresholds are the gold standard for assessing hearing loss, but most clinicians agree that the audiogram must be paired with a speech-in-noise test to make accurate predictions about how listeners will perform in difficult auditory environments. This study evaluated the effectiveness of a six-alternative closed-set speech-in-noise test based on the Modified Rhyme Test (House, 1965). This 104-word test was carefully constructed to present stimuli with and without ITD-based spatial cues at two different levels and two different SNR values. This allows the results to be analyzed not only in terms of overall performance, but also in terms of the impact of audibility, the slope of the psychometric function, and the amount of spatial release from masking for each individual listener. Preliminary results from normal and hearing-impaired listeners show that the increase in overall level from 70 dB to 78 dB that was implemented in half of the trials had little impact on performance. This suggests that the test is relatively effective at isolating speech-in-noise distortion from the effects of reduced audibility at high frequencies. Data collection is currently underway to compare performance in the MRT test to performance in a matrix sentence task in a variety of realistic operational listening environments. [The views expressed in this abstract are those of the authors and do not necessarily reflect the official policy or position of the DoD or the US Government.]

4:30

4pSC12. The contribution of speech motor function to the cognitive testing.

Emily Wang, Stanley Sheft, Valeriy Shafiro (Commun. Disord. and Sci., Rush Univ. Medical Ctr., 1611 West Harrison St., Ste. 530, Chicago, IL 60612, emily_wang@rush.edu), and Raj Shah (The Rush Alzheimer's Disease Core Ctr., Rush Univ. Medical Ctr., Chicago, IL)

This pilot study was to explore speech function as a possible confounding factor in the assessment of persons with Mild Cognitive Impairment (MCI) due to Alzheimer's disease (AD). In the United States, over 30 million people are 65 and older with 10 to 20% of them suffering from MCI due to AD. Episodic memory is tested in diagnosis of MCI due to AD using recall of a story or a list of words. Such tasks involve both speech and hearing. Normal aging also impacts one's speech and hearing. In this study, we designed a test battery to investigate the contribution of speech and hearing on testing of episodic memory. Sixty community-dwelling Black and 60 demographically matched White, all over 74 years, non-demented persons participated in the study. They each produced a story-retell and named animals in one minute. All subjects were tested with hearing and speech measures (maximum-sustained vowel phonation and diadochokinetic rates). Preliminary results showed that small but consistent differences were seen between the two racial groups in the diadochokinetic rates ($p < 0.05$). There were negative correlations between the Story-retell and diadochokinetic rates, which may suggest that speech motor control may indeed be a confounding factor in episodic memory testing.

4:45

4pSC13. The effect of background noise on intelligibility of adults and children with dysphonia.

Keiko Ishikawa (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, 5371 Farmridge Way, Mason, OH 45040, ishikak@mail.uc.edu), Maria Powell (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Amelia, OH), Heidi Phero (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), Alessandro de Alarcon (Pediatric Otolaryngol. Head & Neck Surgery, Cincinnati Children's Hospital Medical Ctr., Cincinnati, OH), Sid M. Khosla (Dept. of Otolaryngol., Univ. of Cincinnati, College of Medicine, Cincinnati, OH), Suzanne Boyce (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH), and Lisa Kelchner (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, 3202 Eden Ave., OH)

A majority of patients with dysphonia report reduced intelligibility in their daily communication environments. Laryngeal pathology often causes abnormal vibration and incomplete closure of the vocal folds, resulting in increased noise and decreased harmonic power in the speech signal. These acoustic consequences likely make dysphonic speech more difficult to understand, particularly in the presence of background noise. The study tested two hypotheses: (1) intelligibility of dysphonic speech is more negatively affected by background noise than that of normal speech, and (2) listener ratings of intelligibility will correlate with clinical measures of dysphonia. One hundred twenty speech samples were collected from 6 adults and 4 children with normal voice and 6 adults and 4 children with varying degrees of dysphonia. Each sample consisted of a short phrase or sentence and was characterized by two acoustic measures commonly associated with degree of dysphonia: cepstral peak prominence (CPP) and harmonic to noise ratio (HNR). Samples were combined with three levels of "cafeteria" noise ($+0$ dB SNR, $+5$ dB SNR, and no noise) and then subjected to a speech perception experiment with 60 normal listeners. This project is ongoing. Preliminary results support hypothesis 1; additional findings related to hypothesis 2 will also be discussed.

Session 4pSP

Signal Processing in Acoustics and Underwater Acoustics: Sensor Array Signal Processing II

Mingsian R. Bai, Chair
Power Mech. Eng., Tsing Hua Univ., 101 Sec.2, Kuang_Fu Rd., Hsinchu 30013, Taiwan

Contributed Papers

1:30

4pSP1. Processing methods for coprime arrays in complex shallow water environments. Andrew T. Pyzdek (Graduate Program in Acoust., The Penn State Univ., PO Box 30, State College, PA 16804, atp5120@psu.edu) and R. Lee Culver (Appl. Res. Lab., The Penn State Univ., State College, PA)

Utilizing the concept of the coarray, coprime arrays can be used to generate fully populated cross-correlation matrices with a greatly reduced number of sensors by imaging sensors to fill in gaps in the physical array. Developed under free space far-field assumptions, such image sensors may not give accurate results in complicated propagation environments, such as shallow water. Taking shallow water acoustic models under consideration, it will be shown that image sensors can still be used, but to a more limited extent based on spatial variability. Performance of a coprime array with limited image sensors and full image sensors will be compared with that of a fully populated array. [This research was supported by the Applied Research Laboratory, at the Pennsylvania State University through the Eric Walker Graduate Assistantship Program.]

1:45

4pSP2. Compressive beamforming in noisy environments. Geoffrey F. Edelmann, Charles F. Gaumond, and Jeffrey S. Rogers (Acoust. (Code 7160), U.S. Naval Res. Lab., 4555 Overlook Ave. SW (Code 7162), Code 7145, Washington, DC 20375, edelmann@nrl.navy.mil)

The application of compressive sensing to detect targets of interest could greatly impact future beamforming systems. Inevitably, at-sea data are contaminated with measured noise. When the ocean is stationary enough to form multiple snap-shots, a covariance matrix may be formed to mitigate noise. Results of compressive beamforming on a covariance matrix will be shown on at-sea measurements. Results will be compared with a robust adaptive beamformer and compressive beamformer. It will be shown that the dictionary of a compressive covariance beamformer goes as the number of measurements squared leading to a compromise between processor and array gain. [This work was supported by ONR.]

2:00

4pSP3. Passive ranging in underwater acoustic environment subject to spatial coherence loss. Hongya Ge (ECE, New Jersey Inst. of Technol., New Jersey Inst. of Technol., University Heights, Newark, NJ 07102, ge@njit.edu) and Ivars P. Kirsteins (Naval Undersea Warfare Ctr., Newport, RI)

In this work, a two-stage multi-rank solution for passive ranging is presented for acoustic sensing systems using multi-module towed hydrophone arrays operating in underwater environments subject to spatial coherence loss. The first stage of processing consists of adaptive beam-forming on the individual modular array level to improve the signal-to-noise ratio and at the same time to adaptively reduce the data dimensionality. The second stage of multi-rank filtering exploits the possible spatial coherence existing across the spatially distributed modular arrays to further improve the accuracy of passive ranging. The proposed solution reduces to either the well-known non-coherent solution under no spatial coherence, or the fully coherent solution under perfect spatial coherence. For large distributed arrays, the asymptotic approximation of the proposed solution has a simple beam-space interpretation. We conclude with a discussion of the estimator when the spatial coherence is unknown and its implications for the passive ranging system performance.

2:15

4pSP4. Eigenvector-based test for local stationarity applied to beamforming. Jorge E. Quijano (School of Earth and Ocean Sci., Univ. of Victoria, Bob Wright Ctr. A405, 3800 Finnerty Rd. (Ring Road), Victoria, BC V8P 5C2, Canada, jorgeq@uvic.ca) and Lisa M. Zurk (Elec. and Comput. Eng. Dept., Portland State Univ. , Portland, OR)

Sonar experiments with large-aperture horizontal arrays often include a combination of targets moving at various speeds, resulting in non-stationary statistics of the data snapshots recorded at the array. Accurate estimation of the sample covariance (prior to beamforming and other array processing procedures) is achieved by including a large number of snapshots. In practice, this accuracy is affected by the requirement to limit the observation interval to snapshots with local stationarity. Data-driven statistical tests for stationarity are then relevant as they allow determining the maximum number of snapshots (i.e., the best case scenario) for sample covariance estimation. This work presents an eigenvector-based test for local stationarity. It can be applied to the improvement of beamforming when targets must be detected in the presence of loud-slow interferers in the water column. Given a set of (possibly) non-stationary snapshots, the proposed approach forms subsets of a few snapshots, which are used to estimate a sequence of sample covariances. Based on the structure of sample eigenvectors, the proposed test gives a probability measure of whether such consecutive sample covariances have been drawn from the same underlying statistics. The approach is demonstrated with simulated data using parameters from the Shallow Water Array Processing (SWAP) project.

2:30

4pSP5. Wind turbine blade health monitoring using acoustic beamforming techniques. Kai Aizawa (Dept. of Precision Mech., Chuo Univ., Lowell, Massachusetts) and Christopher Niezrecki (Dept. of Mech. Eng., Univ. of Massachusetts Lowell, One University Ave., Lowell, MA 01854, Christopher_Niezrecki@uml.edu)

Wind turbines operate autonomously and can possess reliability issues attributed to manufacturing defects, fatigue failure, or extreme weather events. In particular, wind turbine blades can suffer from leading and trailing edge splits, holes, or cracks that can lead to blade failure and loss of energy revenue generation. In order to help identify damage, several approaches have been used to detect cracks in wind turbine blades; however, most of these methods require transducers to be mounted on the turbine blades, are not effective, or require visual inspection. This paper will propose a new methodology of the wind turbine non-contact health monitoring using the acoustic beamforming techniques. By mounting an audio speaker inside of the wind turbine blade, it may be possible to detect cracks or damage within the structure by observing the sound radiated from the blade. Within this work, a phased array beamforming technique is used to process acoustic data for the purpose of damage detection. Several algorithms are

evaluated including the CLEAN-based Subtraction of Point spread function from a Reference (CLSPR) on a composite panel and a section of a wind turbine blade in the laboratory.

2:45

4pSP6. Compressive beamforming with co-prime arrays. Jeffrey S. Rogers, Geoffrey F. Edelmann, and Charles F. Gaumnd (Acoust. Div., Naval Res. Lab, 4555 Overlook Ave. SW, Code 7161, Washington, DC 20375, jeff.rogers@nrl.navy.mil)

The results of compressive beamforming using arrays formed by Nyquist, co-prime samplers, Wichmann rulers, and Golomb rulers are shown along with forms of array gain, resolution and latency as measures of performance. Results will be shown for the idea case of few sources with Gaussian amplitudes in spatially white Gaussian white noise. Results will also be shown for data taken on the Five Octave Research Array (FORA). [This work was supported by ONR.]

3:00

4pSP7. How round is the human head? Buye Xu, Ivo Merks, and Tao Zhang (Starkey Hearing Technologies, 6600 Washington Ave. S, Eden Prairie, MN 55344, buye_xu@starkey.com)

Binaural microphone arrays are becoming more popular for hearing aids due to their potential to improve speech understanding in noise for hearing impaired listeners. However, such algorithms are often developed using three-dimensional head-related transfer function measurements which are expensive and often limited to a manikin head such as KEMAR. As a result, it is highly desired to use a parametric model for binaural microphone array design on a human head. Human heads have been often modeled using a rigid sphere when diffraction of sound needs to be considered. Although the spherical model may be a reasonable model for first order binaural microphone arrays, recent study has shown that it may not be accurate enough for designing high order binaural microphone arrays for hearing aids on a KEMAR (Merks *et al*., 2014). In this study, main sources of these errors are further investigated based on numerical simulations as well as three-dimensional measurement data on KEMAR. The implications for further improvement will be discussed.

3:15–3:30 Break

3:30

4pSP8. Data fusion applied to beamforming measurement. William D. Fonseca (Civil Eng., Federal Univ. of Santa Maria, Rua Lauro Linhares, 657, Apto 203B, FlorianÃpolis, Santa Catarina 88036-001, Brazil, will.fonseca@eac.ufsm.br) and JoÃo P. Ristow (Mech. Eng., Federal Univ. of Santa Catarina, FlorianÃpolis, Santa Catarina, Brazil)

The aim of this work is use data fusion in a set of data obtained from measurements done with a microphone array in different times to improve beamforming results. Beamforming is a technique that basically samples the sound field with an array of sensors. The correct summation of these signals will render a reinforcement of the recorded sound for a chosen direction in space. In addition, processing a set of possible incoming directions enables the creation of sound maps. The spatial resolution in beamforming is directly related to array's constructive factors and frequency of analysis. One way to improve resolution is increasing array's size and number of sensors. Considering the measured source statistically stationary, it is possible to use signals obtained in different times to evaluate it. In this way, the array can be placed in different positions, and the data acquired can be processed and fused in order to create a single set of data corresponding to a virtual array composed by all aforementioned positions.

3:45

4pSP9. Passive multi-target localization by cross-correlating beams of a compact volumetric array. John Gebbie, Martin Siderius (Northwest Electromagnetics and Acoust. Res. Lab., Portland State Univ., 1900 SW 4th Ave., Ste. 160, Portland, OR 97201, jgebbie@ece.pdx.edu), Peter L. Nielsen, and James Miller (Res. Dept., STO-CMRE, La Spezia, Italy)

A technique is presented for passively localizing multiple noise-producing targets by cross-correlating the elevation beams of a compact volumetric array on separate bearings. A target's multipath structure inherently contains information about its range, however unknown, random noise waveforms make time separation of individual arrivals difficult. Ocean ambient noise has previously been used to measure multipath delays to the seabed by cross-correlating the beams of a vertical line array [Siderius *et al*., J. Acoust. Soc. Am. **127**, 2193–2200 (2010)], but this methodology has not been applied to distant noise sources having non-vertical arrivals. In this paper, methods are presented for using a compact volumetric array mounted to an autonomous underwater vehicle to measure the directionality and time delays of multipath arrivals, while simultaneously rejecting clutter and interference. This is validated with results from the GLASS'12 experiment in which a small workboat maneuvered in shallow water. Short ranges could be estimated reliably using straight ray paths, but longer ranges required accounting for ray refraction effects. Further, this is related to striation patterns observed in spectrograms, and it is shown that measured multipath time delays are used to predict this pattern, as well as the waveguide invariant parameter, β.

4:00

4pSP10. Near- and far-field beam forming using a linear array in deep and shallow water. Richard L. Culver, Brian E. Fowler, and D. Chris Barber (Appl. Res. Lab., Penn State Univ., Po Box 30, 16804, State College, PA 16801, r.lee.culver@gmail.com)

Underwater sources are typically characterized in terms of a source level based on measurements made in the free-field. Measurements made in a harbor environment, where multiple reflections, high background noise and short propagation paths are typical, violates these conditions. The subject of this paper is estimation of source location and source level from such measurements. Data from a test conducted at the US Navy Acoustic Research Detachment in Bayview, Idaho during the summers of 2010 and 2011 are analyzed. A line array of omnidirectional hydrophones was deployed from a barge in both deep and shallow water using calibrated acoustic sources to evaluate the effectiveness of post-processing techniques, as well as line array beamforming, in minimizing reflected path contributions and improving signal-to-noise ratio. A method of estimating the location of the sources while taking into account a real, non-linear array based on these measurements is presented. [Work supported by the Applied Research Laboratory under an Eric Walker Scholarship.]

4:15

4pSP11. Two-dimensional slant filters for beam steering. Dean J. Schmidlin (El Roi Analytical Services, 2629 US 70 East, Unit E-2, Valdese, NC 28690-9005, djschmidlin@charter.net)

The concept of a two-dimensional digital "slant" filter is introduced. If the input and output of the slant filter are represented by matrices whose row and column indices denote discrete time and discrete space, respectively, then each diagonal of the output matrix is equal to the linear convolution of the corresponding diagonal of the input matrix with a common one-dimensional sequence. This sequence may be considered as the impulse response of a one-dimensional shift-invariant filter. The transfer function of the slant filter has the form $H(z_1,z_2) = G(z_1z_2)$ where $G(z)$ is the transfer

function of the one-dimensional filter. It is shown that the slant filter is capable of forming and steering a beam using pressure samples from a linear array. The output of the beamformer is equal to the last column of the output matrix of the slant filter. One interesting feature is the possibility that two beamformers can have the same beamwidth but steer the beam to different angles. Another is that though the slant filter is two-dimensional, it can be designed by utilizing well-developed one-dimensional techniques. An example is presented to illustrate the theoretical concepts.

4:30

4pSP12. Compressive acoustic imaging with metamaterials. Yangbo Xie, Tsung-Han Tsai, David J. Brady, and Steven A. Cummer (Elec. and Comput. Eng., Duke Univ., 3417 CIEMAS, Durham, NC 27705, yx35@duke.edu)

Compressive imaging has brought revolutionary design methodologies to imaging systems. By shuffling and multiplexing the object information space, the imaging system compresses data on the physical layer and enables employing fewer sensors and acquiring less data than traditional isomorphic mapping imaging systems. Recently metamaterials have been investigated for designing compressive imager. Metamaterials are engineered materials with properties that are usually unattainable in nature. Acoustic metamaterials can possess highly anisotropy, strongly dispersion, negative dynamic density, or bulk modulus, and they open up new possibilities of wave-matter interaction and signal modulation. In this work, we designed, fabricated, and tested a metamaterial-based single detector, 360 degree field of view compressive acoustic imager. Local resonator arrays are design to resonate randomly in both spatial and spectrum dimensions to favor compressive imaging task. The presented experimental results show that with only about 60 measured values, the imager is able to reconstruct a scene of more than 1000 sampling points in space, achieving a compression ratio of about 20:1. Multiple static and moving target imaging task were performed with this low cost, single detector, non-mechanical scanning compressive imager. Our work paves the way for designing metamaterials based compressive acoustic imaging system.

4:45

4pSP13. Frequency-difference matched field processing in the presence of random scatterers. Brian Worthmann (Appl. Phys., Univ. of Michigan, 2010 W.E.Lay Automotive Lab., 1231 Beal Ave., Ann Arbor, MI 48109, bworthma@umich.edu) and David R. Dowling (Mech. Eng., Univ. of Michigan, Ann Arbor, MI)

Matched field processing (MFP) is an established technique for locating remote acoustic sources in known environments. Unfortunately, unknown random scattering and environment-to-propagation model mismatch prevents successful application of MFP in many circumstances, especially those involving high frequency signals. Recently a novel nonlinear array-signal-processing technique, frequency difference beamforming, was found to be successful in combating the detrimental effects of random scattering for 10 kHz to 20 kHz underwater signals that propagated 2.2 km in a shallow ocean sound channel and were recorded by a 16-element vertical array. This presentation covers the extension of the frequency-difference concept to MFP using sound propagation simulations in a nominally range-independent shallow ocean sound channel that includes point scatterers. Here again, 10 kHz to 20 kHz signals are broadcast to a vertical 16-element array, but the frequency difference approach allows Bartlett and adaptive MFP ambiguity surfaces to be calculated at frequencies that are an order of magnitude (or more) below the signal bandwidth where the detrimental effects of environmental mismatch and random scattering are much reduced. Comparison of these results with equivalent simulations of conventional Bartlett and adaptive MFP for different of source-array ranges are provided. [Sponsored by the Office of Naval Research.]

5:00

4pSP14. Evaluation of a high-order Ambisonics decoder for irregular loudspeaker arrays through reproduced field measurements. Jorge A. Trevino Lopez (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 9808577, Japan, jorge@ais.riec.tohoku.ac.jp), Takuma Okamoto (National Inst. of Information and Communications Technol., Kyoto, Japan), Yukio Iwaya (Faculty of Eng., Tohoku Gakuin Univ., Tagajo, Miyagi, Japan), Shuichi Sakamoto, and Yo-iti Suzuki (Res. Inst. of Elec. Commun. and Graduate School of Information Sci., Tohoku Univ., Sendai, Japan)

High-order Ambisonics (HOA) is a sound field reproduction technique that defines a scalable and system-independent encoding of spatial sound information. Decoding of HOA signals for reproduction using loudspeaker arrays can be a difficult task if the angular spacing between adjacent loudspeakers, as observed from the listening position, is not uniform. In this research, one of such systems is considered: a 157-channel irregular loudspeaker array. The array is used to reproduce simple HOA-encoded sound fields. Three HOA decoding methods are evaluated: two conventional ones and a recently proposed decoder designed for irregular loudspeaker arrays. Reproduction accuracy is compared by directly measuring the sound pressure around the listening position, the so-called sweet spot. Coarse-resolution sound field measurements give an approximate size for the listening region generated by the different methods. In addition, dummy head recordings are used to evaluate interaural level and phase differences. The results are used to estimate the accuracy of the system when presenting spatial sound. This study shows the importance of selecting a proper decoding method to reproduce HOA with irregular loudspeaker arrays. This is emphasized by the use of an actual loudspeakers system instead of a computer simulation, a common shortcoming of previous studies.

Session 4pUW

Underwater Acoustics: Acoustic Vector Sensor Measurements: Basic Properties of the Intensity Vector Field and Applications II

David R. Dall'Osto, Cochair
Acoust., Appl. Phys. Lab. at Univ. of Washington, 1013 N 40th St., Seattle, WA 98105

Peter H. Dahl, Cochair
Appl. Phys.Lab., Univ. of Washington, Mech. Eng., 1013 NE 40th St., Seattle, WA 98105

Invited Papers

1:30

4pUW1. Development of a uniaxial pressure-acceleration probe for diagnostic measurements performed on a spherical sound projector. James A. McConnell (Appl. Physical Sci. Corp. , 4301 North Fairfax Dr., Ste. 640, Arlington, VA 22203, jmcconnell@aphysci.com)

Historically speaking, underwater acoustic vector sensors have seen widespread use in direction finding applications. However, given that this class of sensor typically measures both the acoustic pressure and at least one component of the particle velocity at a single point in space, they can be used effectively to measure the acoustic intensity and/or the acoustic impedance. These metrics can be useful in understanding the acoustic field associated with simple and complex sound radiators. The focus of this paper concerns the development of a uniaxial pressure-acceleration (p-a) probe to measure the specific acoustic impedance of a spherical sound projector (i.e., International Transducers Corporation ITC1001 transducer) over the frequency range from 2.5 to 10 kHz. The design, fabrication, and calibration of the probe are covered along with the results of the aforementioned experiment. Results show that reasonable agreement was obtained between the measured data and an analytical prediction, which models the sound projector as a point source positioned in a free-field.

1:50

4pUW2. An adaptive beamformer algorithm using a quadratic norm of the Poynting vector for vector sensor arrays. Arthur B. Baggeroer (Mech. and Elec. Eng., Massachusetts Inst. of Technol., Rm. 5-206, MIT, Cambridge, MA 02139, abb@boreas.mit.edu)

An adaptive beamformer for vector sensor arrays (VSA's), which uses a quadratic norm of the acoustic Poynting vector (PV) and linear constraint on the PV itself, is introduced. The paradigm follows minimum variance distortionless response (MVDR) but now the metric to be minimized is a quartic function of the filter weights and the constraint is quadratic. This leads to numerical approaches for the optimization instead of a matrix inversion for MVDR. This exploration is motivated by the observation that many nonlinear processing methods lead to "better" performance when a signal is above some threshold SNR. Examples of these include split beam arrays, DIFAR's and monopulse systems. This presentation discusses the optimization method and compares the results for ABF with linear processing for VSA's. The use of linear and quadratic refer to the clairvoyant processing where the ABF uses ensemble covariances and leaves open the problem of sample covariance estimation. [Work supported by ONR Code 321, Undersea Signal Processing.]

Contributed Papers

2:10

4pUW3. The modal noise covariance matrix for an array of vector sensors. Richard B. Evans (Terrafore. Inc., 99F Hugo Rd., North Stonington, CT 06359, rbevans@99main.com)

A modal noise covariance matrix for an array of vector sensors is presented. It is assumed that the sensors measure pressure and gradients or velocities on three axes. The noise covariance matrix is obtained as a discrete modal sum. The derivation relies on the differentiation of the complex pressure field and the application of a set of Bessel function integrals. The modal representation is restricted to a horizontally stratified environment and assumes that the noise sources form a layer of uncorrelated monopoles. The resulting noise field is horizontally isotropic, but vertically non-isotropic. Particular attention is paid to the effect of the noise source intensity on the normalization of the covariance matrix and, consequently, to the effect of noise on the output of the array of vector sensors.

2:25

4pUW4. Bearing estimation from vector sensor intensity processing for autonomous underwater gliders. Kevin B. Smith, Timothy Kubisak, James M. Upshaw (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Rm. 114, Monterey, CA 93943, kbsmith@nps.edu), James S. Martin, David Trivett (Woodruff School of Mech. Eng., Georgia Inst. of Technol. , Atlanta, GA), and C. Michael Traweek (Office of Naval Res., Arlington, VA)

Data have been collected on acoustic vector sensors mounted on autonomous underwater gliders in the Monterey Bay during 2012–2013. In this

work, we show results of intensity processing to estimate bearing to impulsive sources of interest. These sources included small explosive shots deployed by local fisherman, and humpback whale vocalizations. While the highly impulsive shot data produced unambiguous bearing estimations, the longer duration whale vocalizations showed a fairly wide spread in bearing. The causes of the ambiguity in bearing estimation are investigated in the context of the highly variable bathymetry of the Monterey Bay Canyon, as well as the coherent multipath interference in the longer duration calls.

2:40

4pUW5. Detection and tracking of quiet signals in noisy environments with vector sensors. Donald DelBalzo (Marine Information Resources Corp., 18139 Bellezza Dr., Orlando, Florida 32820, delbalzo@earthlink.net), James Leclere, Dennis Lindwall, Edward Yoerger, Dimitrios Charalampidis, and George Ioup (Phys., Univ. of New Orleans, New Orleans, LA)

We analyze the utility of vector sensors to detect and track underwater acoustic signals in noisy environments. High ambient noise levels below 300 Hz are often dominated by a few loud discrete ships that produce a complicated and dynamic noise covariance structure. Horizontal arrays of omni-directional hydrophones improve detection by forming (planewave) beams that "listen" between loud azimuthal directions with little regard to changing noise fields. The inherent 3-D directionality of vector sensors offers the opportunity to exploit detailed noise covariance structure at the element level. We present simulation performance results for vector sensors in simple and realistic environments using particle filters that can adapt to changing acoustic field structures. We demonstrate the ability of vector sensors to characterize and mitigate the deleterious effects of noise sources. We also demonstrate the relative value of vector vs. omni-directional sensing (and processing) for single sensors and compact arrays.

2:55

4pUW6. Coherent vector sensor processing for autonomous underwater glider networks. Brendan Nichols, James Martin, Karim Sabra, David Trivett (Mech. Eng., Georgia Inst. of Technol., 801 Ferst Dr. NW, Atlanta, GA 30309, bnichols8@gatech.edu), and Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, Monterey, CA)

A distributed array of autonomous underwater gliders, each fitted with a vector sensor measuring acoustic pressure and velocity, form an autonomous sensor network theoretically capable of detecting and tracking objects in an ocean environment. However, uncertainties in sensor positions impede the ability of this glider network to perform optimally. Our work aims to compare the performance of coherent and incoherent processing for acoustic source localization using an array of underwater gliders. Data used in the study were obtained from numerical simulations as well as experimental data collected using the research vessel as a source for localization purposes. By estimating the vessel position with a single glider's data (incoherent) and comparing to the location estimated with both gliders' data (coherent), it was determined that location estimation accuracy could be improved using coherent processing, provided the gliders' positions could be measured with sufficient precision. The results of this study could potentially aid the design and navigation strategies of future glider networks with a large number of elements.

3:10–3:30 Break

3:30

4pUW7. Development of vector sensors for flexible towed array. Vladimir Korenbaum and Alexandr Tagiltcev (Pacific Oceanologic Inst. FEB RAS, 43, Baltiiskaya Str., Vladivostok 690041, Russian Federation, v-kor@poi.dvo.ru)

Main problems of application of vector sensors (VSs) for flexible towed arrays are providing high performance under small dimensions as well as necessary flow noise immunity. The objective is to develop VSs met these demands. A simulation of performance of VS embedded in a flexible towed array body formed with sound transparent compound is performed. The developed one-dimensional model, predicts existence of a suspension resonance, dividing frequency band of VS into two parts. The lower part of the band is more applicable for VS of inertial type while the upper one is more preferred for VS of gradient type. A possibility to control the suspension resonance frequency in limits of 500–2000 Hz is shown for experimental model. The flow noise immunity problem is analyzed for different frequency bands and types of VSs. Various methods of flow noise cancellation are developed for different frequency bands and types of VSs, which include power flux processing, compensation of vibration response, convolution processing. Examples of design of one- and two-component VSs are represented. [The study was supported by the grant 13-NTP-II-08 of Far Eastern Branch of Russian Academy of Sciences.]

3:45

4pUW8. Acoustic particle velocity amplification and flow noise reduction with acoustic velocity horns. Dimitri Donskoy (Ocean Eng., Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, ddonskoy@stevens.edu) and Scott E. Hassan (Naval Undersea Warfare Ctr., Newport, RI)

Small wavelength size acoustic velocity horns (AVH) were recently introduced [J. Acoust. Soc. Am. **131**(5), 3883–3890 (2012)] as particle velocity amplifiers having flat amplitude and phase frequency responses below their first resonance. AVH predicted amplification characteristics have been experimentally verified demonstrating interesting opportunities for vector sensors (VS) sensitivity enhancement. Present work provides enhanced analysis of amplification and characteristics of complex shape horns. Additionally, we address another AVH feature: turbulence flow noise reduction due to turbulence field spatial averaging across horn's mouth area. Numerical analysis demonstrated up to 25 dB convective turbulent pressure and velocity reduction at the horn throat.

4:00

4pUW9. Development of a standing-wave calibration apparatus for acoustic vector sensors. Richard D. Lenhart, Jason D. Sagers (Appl. Res. Lab., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, lenhart@arlut.utexas.edu), and Preston S. Wilson (Mech. Eng. Dept. and Appl. Res. Lab., The Univ. of Texas at Austin, Austin, TX)

An apparatus was developed to calibrate acoustic hydrophones and vector sensors between 25 and 2000 Hz. A standing-wave field is established inside a vertically oriented, water-filled, elastic-walled waveguide by a piston velocity source at the bottom and a pressure-release boundary condition at the air/water interface. A computer-controlled linear positioning system allows reference hydrophones and/or the device under test to be scanned through the water column while their acoustic response is measured. Some of the challenges of calibrating vector sensors in such an apparatus are discussed, including designing the waveguide to mitigate dispersion, mechanically isolating the apparatus from floor vibrations, understanding the impact of waveguide structural resonances on the acoustic field, and developing processing algorithms to calibrate vector sensors in a standing-wave field. Data from waveguide characterization experiments and calibration measurements will be presented. [Work supported by ARL IR&D.]

4:15

4pUW10. Very low frequency acoustic vector sensor calibration. Dimitri Donskoy (Ocean Eng., Stevens Inst. of Technol., 711 Hudson St., Hoboken, NJ 07030, ddonskoy@stevens.edu)

In-water calibration of Acoustic Vector Sensors (AVS) operating at very low frequencies (fraction of Hz to hundreds of Hz) presents a set of unique challenges as the acoustic wavelengths are much longer than any existing laboratory calibration facilities. The developed calibration approach utilizes existing Naval Undersea Warfare Center's pressurized horizontal calibrating steel tube equipped with two independently controlled sound sources located at the opposite ends of the tube. Controlling the phase and amplitude of these sources allows for creating of pressure or velocity fields inside the tube. Respective pressure and particle velocity complex amplitudes are measured and calculated, respectively, with two reference hydrophones. Experimental results of this calibration approach is presented for a newly developed very low frequency AVS comprising of pressure and non-inertial velocity sensors built into an acoustic velocity horn.

4:30

4pUW11. Spatial correlation of the acoustic vector field of the surface noise in three-dimensional ocean environments. Yiwang Huang and Junyuan Guo (College of Underwater Acoust. Eng., Harbin Eng. Univ., Nantong St. No.145, Nangang District, Heilongjiang,Harbin 150001, China, guojunyuan89@163.com)

Spatial correlation of ocean ambient noise is a classical and attractive topic in ocean acoustics. Usually acoustic particle velocity can be formulated by the gradient of sound pressure. But due to the complexity of the sound pressure in range-dependent environments, the velocities of the surface noise are too difficult to be solved by this way. Fortunately, by taking advantage of the exchangeability of partial derivative and integral operation, a new derivation was proposed and a vector model for the surface-generated noise in three-dimensional ocean environments was developed directly from the correlation function of sound pressure. As a model verification, spatial correlation of the acoustic vector field of the surface noise in a range-independent environment was derived, and the identical correlation functions were given compared with the literature. After then, the surface noise in a range-dependent environment was considered with a rigid bottom hypothesis. The effects on the correlation taken by the bottom sloping and medium absorption were analyzed numerically.

THURSDAY EVENING, 8 MAY 2014 — 7:30 P.M. TO 9:30 P.M.

OPEN MEETINGS OF TECHNICAL COMMITTEES

The Technical Committees on the Acoustical Society of America will hold open meetings on Tuesday, Wednesday and Thursday evenings.

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:

7:30 p.m.	Animal Bioacoustics	554AB
7:30 p.m.	Biomedical Acoustics	Ballroom E
7:30 p.m.	Musical Acoustics	Ballroom C
7:30 p.m.	Noise	557
7:30 p.m.	Speech Communication	Ballroom D
7:30 p.m.	Underwater Acoustics	556AB

Session 5aAAa

Architectural Acoustics and Psychological and Physiological Acoustics: Psychoacoustics in Rooms II

Philip W. Robinson, Cochair
Media Technol., Aalto Univ., PL 15500, Aalto 00076, Finland

Frederick J. Gallun, Cochair
National Ctr. for Rehabilitative Auditory Res., Portland VA Medical Ctr., 3710 SW US Veterans Hospital Rd., Portland, OR 97239

Chair's Introduction—8:00

Invited Papers

8:05

5aAAa1. Some effects of room acoustics and background noise on assistive listening systems and devices. Peter Mapp (Peter Mapp Assoc., Copford, Colchester CO6 1LG, United Kingdom, petermapp@petermapp.com)

Approximately 10–14% of the general population (United States and Northern Europe) suffer from a noticeable degree of hearing loss and would benefit from some form of hearing assistance or deaf aid. However, many hearing aids do not provide a suitable level of intelligibility when used in large reverberant or noisy spaces and rooms. The paper investigates the acoustic and speech intelligibility requirements for Assistive Listening Systems that may be used separately or in conjunction with a hearing aid to improve the potential intelligibility of the received speech signal. A number of microphone pick-up scenarios have been investigated and are reported in terms of their potential intelligibility and sound quality performance. The results of testing carried out in a number of rooms and venues are presented, mainly in terms of the resultant Speech Transmission Index (STI). The paper concludes by providing a number of recommendations and guidelines for successful microphone placement and introduces a novel technique for establishing useful talker sound radiation and hence microphone target aiming.

8:25

5aAAa2. Investigation of subjective components of overall acoustic quality using binaural recordings made in Hartford's Belding Theater. Acadia A. Kocher (Northwestern Univ., 6120 Holly Ridge Ct, Columbia, Maryland 21044, acadiakocher2015@u.northwestern.edu) and Michelle C. Vigeant (Graduate Program in Acoust., The Penn State Univ., University Park, PA)

Previous research in concert hall acoustics has established correlations between listener preference for overall acoustic quality (OAQ) and a number of other subjective factors. A subjective study was conducted to determine the relative importance of several characteristics to OAQ. Subjects evaluated 18 signals based on listener envelopment, reverberance, tonal quality, and overall preference using five-point rating scales. The signals were presented individually over headphones. A total of 32 subjects with formal musical training and normal hearing were included in the study. Following a brief tutorial and training session, the test was divided into four sets, one for each of the subjective factors. All of the signals and sets were presented to each subject in a random order. The binaural recordings were made in the Belding Theater in Hartford, Connecticut. A loudspeaker on the stage played three short classical motifs that were recorded in three seat locations with two distinct settings of the hall's variable acoustics system. The sound pressure levels of all signals were normalized to isolate the characteristics of interest. The data analysis identified relationships between the three factors and OAQ, with the most significant relationship between OAQ and tonal quality. [Work was supported by NSF Grant 1302741.]

8:45

5aAAa3. Localizing low-frequency sounds in a room and the dominance of interaural level differences. William M. Hartmann, Brad Rakerd, Eric J. Macaulay, and Zane D. Crawford (Michigan State Univ., 567 Wilson Rd., East Lansing, MI 48824, hartman2@msu.edu)

In a room-environment study of the duplex theory of sound localization, listeners reported the azimuthal locations of low-frequency sine tones in free field and in three very different rooms. Interaural differences in time (ITD) and level (ILD) were continuously monitored by probe microphones in the listeners' ear canals. As the frequency increased from 250 to 1000 Hz, the correlation of listener responses with the ILD increased while the correlation with the ITD decreased precipitously. The increased importance of the ILD was especially prominent for the least reverberant environments. The decreased importance of the ITD occurred primarily because of interaural phase differences (IPD) that became large (>90 degrees) and consequently weak. The large IPD effect occurred more frequently in the highly reverberant room, and in the less reverberant rooms at 750 and 1000 Hz, where the peak of the IPD distribution occurred well above 90 degrees. The two effects caused the ILD to become more important than the ITD when the frequency was greater than about 500 Hz with only a small dependence on the different rooms. The increased emphasis given to the ILD normally led to more accurate localization. [Work supported by AFOSR grant FA9550-11-1-0101.]

9:05

5aAAa4. On the sensitivity of older hearing-impaired individuals to acoustic attributes. William M. Whitmer and Michael A. Akeroyd (Scottish Section, MRC/CSO Inst. of Hearing Res., Glasgow Royal Infirmary, Glasgow, United Kingdom, bill@ihr.gla.ac.uk)

In previous work, we have run a series of experiments with hearing-impaired adults to examine how age and hearing loss affect sensitivity to changes in apparent auditory source width. In two experiments, the interaural coherence of broadband noises presented over headphones and loudspeakers was varied to induce changes in width; in a third, older participants sketched the width of noise, speech, and musical stimuli in simulated rooms with varying reflection absorption. The results of those experiments showed generally decreasing sensitivities to interaural-coherence-induced changes in width as a function of age and hearing impairment. To examine how these results might influence acoustic design for the aged, a new study considers a more basic task that avoided the auditory-visual transformation implicit in sketching. Participants with normal to moderately impaired hearing will compare simulated utterances in a same/different room-discrimination task. Binaural impulse responses will be generated for open-plan buildings of varying size from the ODEON database with sources and receivers centered in each space, convolved with speech and music tokens and presented over headphones. The differences in the relationship between presbycusis and sensitivity to source width vs. general acoustical attributes will be discussed. [Work supported by the MRC and the CSO.]

9:25

5aAAa5. Amplitude modulation sensitivity in rooms. Pavel Zahorik (Heuser Hearing Inst. and Div. of Communicative Disord., Dept. of Surgery, Univ. of Louisville School of Medicine, Univ. of Louisville, Louisville, KY 40292, pavel.zahorik@louisville.edu)

The physical effects of room acoustics on sound emitted from an amplitude-modulated (AM) source is well understood and can be effectively characterized by the modulation transfer function (MTF) of the room. Human sensitivity to AM is also well understood under conditions which minimize the acoustical contributions of the listening environment, and can analogously be characterized using the MTF concept. Although it may be predicted that AM sensitivity in a reverberant soundfield could be explained if both AM sensitivity in the absence of the soundfield and the MTF of the room are known, little empirical data exist in the literature to validate this prediction. Here, we summarize recent work in our laboratory that addresses this issue. A consistent finding is that predicted AM sensitivity underestimates measured sensitivity for reverberant soundfield listening. This result appears to depend critically on prior listening exposure to the soundfield, and is consistent with recent neural data from other laboratories. It may also explain why speech understanding in normal-hearing populations is typically unaffected by the acoustical degradations caused by reverberation. [Work supported by NIH-NIDCD.]

9:45

5aAAa6. Meeting the classroom acoustics standard on a historical room. Ana M. Jaramillo (Olson Sound Design, 3711 Lake Dr., 55422, Robbinsdale, MN 55422, ana.jaramillo@afmg.eu) and Bruce C. Olson (Olson Sound Design, Brooklyn Park, MN)

The Department of Speech-Language-Hearing Sciences at University of Minnesota occupies Shevlin Hall built in 1906. We were called in 2013 to help with the acoustic and sound system redesign of room 110 to be used as a classroom, as the room had very poor speech intelligibility due to the very high ceiling and hard walls, resulting in a very reverberant environment. Our goal was to meet the current classroom acoustics standard ANSI S12.60. Reverberation time and noise measurements were performed, and the room was modeled using EASE for the prediction of results. The model was compared with measurements, and several alternatives were explored. After the changes were implemented, measurements confirmed the improvement to the room's intelligibility was acquired and the historic character of the room was preserved.

10:05–10:15 Break

Contributed Papers

10:15

5aAAa7. Perception of loudness in directional sound fields. Philip W. Robinson (Media Technol., Aalto Univ., PL 15500, Aalto 00076, Finland, philrob22@gmail.com) and Jukka Pätynen (Media Technol., Aalto Univ., Espoo, Finland)

An often utilized assumption in room acoustics is that the room produces a diffuse field, in which sound is uniformly distributed in space and arrives at the listener equally from all directions. This assumption greatly simplifies many calculations, e.g., Sabine's reverberation time (RT). However, the reverberant field, particularly the early part of the impulse response in typical concert spaces, is highly directional. Hence, the diffuse assumption leads to errors, for example, in the prediction of loudness when using omni-directional measures. Lateral reflections contribute to perceived loudness more than their omni-directionally measured sound energy. This is due the filtering of the head and torso, which amplify reflections from lateral directions more than others, particularly at high frequencies. Listening test results will be presented that demonstrate this effect. Spatial analyses of concert hall impulse responses demonstrate the practical applicability of this finding and make evident the relevance of the effect.

10:30

5aAAa8. Temporal weighting of binaural cues in real rooms: Psychophysical and neural modeling. G. Christopher Stecker (Hearing and Speech Sci., Vanderbilt Univ., 1215 21st Ave. South, Rm. 8310, Nashville, TN 37232, g.christopher.stecker@vanderbilt.edu), Andrew D. Brown, and Daniel J. Tollin (Dept. of Physiol. and Biophys., Univ. of Colorado School of Medicine, Denver, CO)

Although interaural time and level differences (ITD and ILD) comprise the primary cues to sound location in azimuth, both are distorted by echoes and reverberation in many real environments. Consequently, the "effective" cues exhibit complexities and evolve temporally due to interactions of direct and reflected sound. One approach to studying the perceptual effects of such distortions is to measure the relative influence, or "temporal weight," of cues contained within brief temporal segments of sound on spatial judgments made by human listeners. Temporal weighting functions (TWFs) measured in this way reveal binaural sensitivity to be temporally nonuniform and cue-specific. For many sounds (tones and high-rate click trains), judgments appear dominated by the ITD and ILD occurring at sound onset and the ILD occurring near sound offset, while middle portions contribute very little, consistent with expectations about the temporal statistics of these cues in real rooms. In this presentation, psychophysically derived TWFs are compared to the frequency-dependent statistics of ITD and ILD in room recordings. Models of the auditory periphery and early nervous system are used to transform the recordings to estimate the effective cues "as heard by" central brain mechanisms. [Work supported by NIH R01 DC 011548 (GCS) and DC 011555 (DJT).]

10:45

5aAAa9. Designing an auditory lab including active acoustics. Ranil Sonnadara (McMaster Inst. for Music and the Mind, McMaster Univ., Hamilton, ON, Canada), Steve Ellison (Meyer Sound Labs, Inc., 2832 San Pablo Ave., Berkeley, CA 94702, ellison@meyersound.com), Laurel Trainor, and Dan Bosnyak (McMaster Inst. for Music and the Mind, McMaster Univ., Hamilton, ON, Canada)

LIVELab, a new facility at McMaster University, has been developed in order to help support research in areas such as developing intelligent hearing aids, measuring the physiological impact of media presentations, understanding interactions between musicians, and determining effective sounds for medical and warning systems. This room combines a quiet performance space featuring a low nominal reverberation time with technology including active acoustics, multichannel playback, and integral measurement and instrumentation to create a sonically flexible facility for auditory research. The design of the room and its acoustic variability for supporting these research endeavors will be discussed.

11:00

5aAAa10. Perceptually evaluating ambisonic reproduction of room acoustics. Samuel Clapp (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St., Greene Bldg., Troy, NY 12180, clapps@rpi.edu), Anne Guthrie (Arup Acoust., New York, NY), Jonas Braasch, and Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., Troy, NY)

Spherical microphone array technology allows for the recording of auditory scenes in three dimensions by decomposing a soundfield into its spherical harmonic components. The performance of the array is affected by certain factors in its design, including the size of the array, whether an open or a rigid sphere configuration is used, and the number of sensors and their placement scheme. Ambisonics is a system designed to reconstruct a soundfield from its spherical harmonic components. The process of ambisonic decoding determines the signals that are fed to each loudspeaker in the array to simulate a given soundfield. Such systems have certain accuracy constraints, particularly at higher frequencies, and different decoding methods can be used at those frequencies to recreate more accurate spatial cues, particularly ILD cues. This paper examines how to develop a decoding scheme that addresses the constraints in both the recording and playback phases, and uses binaural modeling to determine its efficacy. Reconstruction of both simulated and real rooms is examined, with respect to the accurate reproduction of important room acoustic metrics.

11:15

5aAAa11. Subjective perception of varying reflection densities in room impulse responses. Hyun Hong and Lily M. Wang (Durham School of Architectural Eng. and Construction, Univ. of Nebraska - Lincoln, 1110 S. 67th St., Omaha, NE 68182-0816, hhong@huskers.unl.edu)

The commonly used objective metrics for analyzing room acoustics are each tied to some aspect of subjective perception. For example, the widely used reverberation time is linked to the perceived reverberation in a room. Two different rooms having the same reverberation time, though, can have different reflection densities in their impulse responses, and this difference in reflection density may affect how listeners perceive the size of the rooms. This project investigates the subjective perception of reflection densities and how sensitive humans are to a change of reflection density. First, assorted parameters for quantifying reflection density are reviewed. Then, preliminary results of perceptual tests using different room impulse responses with varying reflection densities are presented.

11:30

5aAAa12. Influence of manipulated early reflections on room acoustical perception. Stefan Klockgether and Steven van de Par (Appl. Phys. and Acoust., Carl von Ossietzky Univ., Carl-von-Ossietzky-Str. 9-11, Oldenburg, Lower Saxony 26129, Germany, stefan.klockgether@uni-oldenburg.de)

The binaural room impulse response (BRIR) can be used to study the perception of room acoustics and consists of different parts such as the direct sound, early and late reflections, and a diffuse reverberant tail. For this study, the contribution of early reflections to several perceptual attributes was investigated. For this purpose, the strength of the early reflections of recorded BRIRs were either increased or decreased. The resulting BRIRs as well as the original BRIRs have then been convolved with anechoic music signals to obtain the stimuli that were presented to the subjects in a psychoacoustic experiment. The subjects had to rate the spatial impression of these manipulated and none-manipulated signals for the perceptual attributes "Listener envelopment,"" Apparent source width," "Distance" and "Presence." In addition to determining the influence of manipulated early reflections on the spatial impression of a room, also a comparison was made with the perceptual effect of an artificial increase of the interaural cross-correlation. Results indicate that the perceived source width increases with increasing level of the early reflections. The effect of the level manipulations on the listener envelopment seems to be small compared to the influence of the cross-correlation.

Session 5aAAb

Architectural Acoustics: Exploring the 2014 Sound and Vibration Guidelines and Case Studies for Healthcare Facilities

Kenric D. Van Wyk, Cochair
Acoustics By Design, Inc., 124 Fulton St. East, Second Fl., Grand Rapids, MI 49503

Daniel M. Horan, Cochair
Cavanaugh Tocci Associates, Inc., 327 F Boston Post Rd., Sudbury, MA 01776

Edward Logsdon, Cochair
D. L. Adams Associates, Inc., 1536 Ogden St., Denver, CO 80218

Chair's Introduction—8:00

Invited Papers

8:05

5aAAb1. Changes in the acoustical requirements of the 2014 editions of the Facility Guidelines Institute's *Guidelines for Design and Construction of Hospitals and Outpatient Facilities* and its reference document *Sound & Vibration Design Guidelines for Health Care Facilities*. Daniel M. Horan (Cavanaugh Tocci Assoc., Inc., 327 F Boston Post Rd., Sudbury, MA 01776, dhoran@cavtocci.com)

Acoustical design criteria for health care facilities as defined by the FGI *Guidelines* have been updated in the recently published 2014 edition. *Sound & Vibration Design Guidelines for Health Care Facilities* (S&V3.0) serves as the sole acoustical reference for the FGI *Guidelines* and has also been updated as part of the 2014 FGI cycle. The S&V reference is authored and edited by the FGI's acoustical working group. The secretary of this group (Horan) will summarize these changes and will also discuss the public review and comment process that led to the updated *Guidelines*. This same process will be used in the forthcoming cycle in preparation for the 2018 edition(s).

8:25

5aAAb2. Evidence based design for improved patient experience. Melinda Miller (Acoustics By Design, Grand Rapids, MI), Kenric Van Wyk, and Kristen Murphy (Acoustics By Design, 124 Fulton St. E, Grand Rapids, MI 49503, kvanwyk@acousticsbydesign.com)

In addition to the health benefits of improving acoustical comfort for patients and staff in healthcare environments, there is now a financial incentive. Since October 2012, Value Based Purchasing reimbursements of Medicare from the Federal Government is dependent in part upon Hospital Consumer Assessment of Healthcare Providers and Systems (HCAHPS) patient satisfaction survey results, which includes a rating for the noise level at night. This study explores the relationship of sound level with their associated patient satisfaction scores from hospital noise surveys performed by Acoustics By Design. The noise study results are presented across different parameters such as the type of patient unit, patient noise levels, nurses' station noise levels, time of day, and private versus semi-private rooms. Measurements were also performed prior to and after completion of acoustical/noise mitigation.

8:45

5aAAb3. Modular construction in healthcare. Edward Logsdon and Benjamin Seep (Acoustical Consulting, D. L. Adams Assoc., Inc., 1536 Ogden St., Denver, CO 80218, elogsdon@dlaa.com)

Healthcare facilities are utilizing modular constructions where elements like the patient restroom or headwall are built off-site and then shipped to the project where they are then installed. The modular assemblies are constructed under tight quality control and the manufacturers suggest savings in labor, shortened schedules, and improved architectural and acoustical performance. Modular patient restroom assemblies were used in the construction of a new major Colorado hospital with 360 private patient rooms. This includes the restrooms used in the labor and delivery areas of the hospital. Acoustical plus and minuses with the use of the modular patient restrooms will be discussed along with recommendations to maintain sound isolation between the spaces. This includes detailing during design and modifications needed to address issues in the field.

9:05

5aAAb4. The implication of door undercut in patient room to corridor speech privacy. Gregory C. Tocci (Cavanaugh Tocci Assoc., Inc., 327F Boston Post Rd., Sudbury, MA 01776, gtocci@cavtocci.com)

Un-gasketted corridor door undercuts and un-gasketted head, hinge, and lockset jambs compromise the STCc rating of corridor wall/door assemblies. However, health care infection control policies often do not permit gasket systems on many doors. Though loss of sound isolation is obvious, this paper specifically characterizes the loss of speech privacy between patient rooms and corridors with doors not provided with undercut drop seals and jamb gaskets. The work draws upon that of Kim and An, and characterizes speech privacy conditions using the articulation index (AI) and its compliment the privacy index (PI).

9:25

5aAAb5. The effect of implementing an electronic sound masking system into a 42-bed oncology unit on "quiet at night" Hospital Consumer Assessment of Healthcare Providers and Systems (HCAHPS) scores. Gary Madaras (Acoust., Making Hospitals Quiet | Rockfon, 4849 S. Austin Ave., Chicago, IL 60638, DoctorSonics@aol.com)

In December 2012, the facilities personnel at a major New York medical center used a donation to install an electronic sound masking system in the patient rooms of a 42-bed oncology care unit. Monthly top-box scores for the unit's 'quiet at night' HCAHPS question as reported by Press Ganey for the years preceding and succeeding the installation will be provided and discussed. Other observations based on staff interviews, personal listening, and acoustical measurements will be provided.

9:45–10:00 Break

10:00

5aAAb6. Integrated design experience and post-occupancy acoustic performance of medical office building of the future. Erik Miller-Klein (SSA Acoust., LLP, 222 Etruria St, Ste. 100, Seattle, WA 98109, erik@ssaacoustics.com)

This case study discusses the experiences working within this integrated design team with architects, engineers, and contractors from day one, which included mock-ups and physical models of proposed systems. This project was completed in 2013, and was designed and evaluated compared to the 2010 Facilities Guidelines Institute Guidelines for Design and Construction of Health Care Facilities. During construction, we evaluated some acoustic construction defects and addressed these impacts, and completed a full acoustic evaluation of the space prior to occupancy. This building was a paradigm shift for the doctors and nurses, and required additional testing and analysis to interpret some of the feedback from users about the acoustic function of the spaces.

10:20

5aAAb7. An approach for estimating noise induced hearing loss because of indoor and outdoor environment noise factors in healthcare facilities. Filiz B. Kocyigit (Architecture, Atilim Univ., Kizilcasar Mah. Incek, Ankara 06560, Turkey, filizbk@gmail.com)

This paper investigates a number of healthcare centers from University Medical Schools and State Hospitals, Baskent University Medical School and Private Research Hospitals, and the State Hospital. The sound levels present in these centers were compared with number international as well as Turkish standards. Research was carried out in the following way: First, physicians and nurses were surveyed by means of a questionnaire. Then, the hospital buildings were examined in terms of their architectural design features. Lastly, sound levels were measured in the consequent spaces where questionnaires were also conducted. In addition, noise in hospitals can be detrimental as it helps to present their environment as quiet and peaceful. WHO provides guidelines for hospitals in this respect in its Guidelines for Community Noise published in 1995 (4). ANSI S12.2, recommends a maximum Room Criterions Curves (NCC) value ranging on the room type, and a maximum Noise Criteria Balanced (NCB) value ranging. A document issued by the Environmental Protection Agency (EPA) summarizing significant community noise studies provides recommendations in terms of the Ld&n (day-night sound pressure level). The Institute of Turkish Standards (TSE) is also working on this subject and the first national standard about ambient at Table 11.

10:40

5aAAb8. A review of hospital noise studies. Melinda Miller, Kenric Van Wyk, and Kristen Murphy (Acoustics By Design, 124 Fulton St. E, Grand Rapids, MI 49503, melinda@acousticsbydesign.com)

Hospital noise concerns are on the rise due to detrimental health effects and the implementation of Value Based Purchasing, which affects federal reimbursements based on HCAHPS patient satisfaction scores. Because noise is one of the lowest performing categories in the survey, it is expected that demand for hospital noise surveys will increase. Therefore, it is useful to examine existing hospital noise measurement techniques to serve as a guideline for future work. This paper provides a summary of several noise studies of existing healthcare facilities conducted by Acoustics By Design (ABD) and others. The goal of the studies performed by ABD was to balance providing a comprehensive noise study (across multiple noise variation factors, such as time of day, and weekend and weekday conditions, department type, etc.) with providing a study that is time and cost efficient. We have found that this is best done through a combination of short term and long term measurements in close collaboration with hospital staff and administration.

Contributed Papers

11:00

5aAAb9. Trends of the acoustic condition in an intensive care unit based on a long-term measurement. Munhum Park (Smart Sensing & Anal., Philips Res. Labs., High Tech Campus 36.p.078, Eindhoven 5656AE, Netherlands, mun.park@philips.com), Piet Vos (Dept. of Intensive Care, St. Elisabeth Hospital, Tilburg, Netherlands), Armin Kohlrausch (Smart Sensing & Anal., Philips Res. Labs., Eindhoven, Netherlands), and Annemarie W. Oldenbeuving (Dept. of Intensive Care, St. Elisabeth Hospital, Tilburg, Netherlands)

Noise levels in hospitals, especially in intensive care units (ICUs), are often very high, potentially influencing the patients' well-being and recovery processes, where the undesirable acoustic environment is also considered to be one of the risk factors contributing to ICU delirium. In the current study, a continuous measurement was taken for 3 months in 8 single-bed patient rooms in an ICU, of which the results were analyzed in synchrony with the admission of 106 patients. On average, the A-weighted energy-equivalent sound pressure level (L_{Aeq}) in patient rooms varied significantly with the time of day ($p < 0.001$), but was not dependent on the day of week ($p = 0.448$). Furthermore, analysis of noise levels in occupied versus unoccupied rooms indicated the dominance of room-internal sources in the former and room-external sources in the latter periods. During the first four days of patients' ICU stay, the acoustic condition improved slightly from day 1 to day 2, but the noise level rebounded from day 2, most likely in relation to the various phases of treatment and recovery. Between-patient variability was found to be significant, which may be an important aspect to consider when comparing the acoustic conditions in an environment occupied by different groups of patients.

11:15

5aAAb10. Study to optimize speech clarity in a hospital pediatric trauma room, using newly patented tuning tube and a custom nonwoven fabric. Bonnie Schnitta (SoundSense, LLC, 46 Newtown Ln., Ste. One, East Hampton, NY 11937, bonnie@soundsense.com)

There are several acoustic requirements in order to optimize the healing environment for the patients in a trauma room, as well as increase medical staff concentration and clarity of speech within the room. This is especially true in a pediatric setting. First, there is the requirement that the reverberation time of the room must be reduced in order to assist in hearing/speech clarity while adhering to clean room standards. Secondly, there needs to be an increase in the signal to noise ratio (SNR). A greater SNR assists in hearing/speech clarity; the lower reverberation time also assists in preventing the various intrusive mechanical sounds from becoming amplified and further reducing the SNR. In addition to these two standard room requirements, there is also a need for the reduction of lower frequency sounds within a room, such as sounds typical of mechanical equipment. Since standard products used to absorb sound have more absorption in speech frequencies, there is a need for products that have greater absorption in lower frequencies. This paper presents background on two SoundSense patented products that were used in a study for improved conditions and outcomes in a hospital pediatric trauma room.

11:30–12:00 Panel Discussion

FRIDAY MORNING, 9 MAY 2014 — 554 A/B, 8:55 A.M. TO 11:00 A.M.

Session 5aAB

Animal Bioacoustics and Education in Acoustics: Communicating the Science of Underwater Sound

Kathleen J. Vigness-Raposa, Chair
Marine Acoustics, Inc., 809 Aquidneck Ave., Middletown, RI 02842

Chair's Introduction—8:55

Invited Papers

9:00

5aAB1. The science of underwater sound: Education, communication, and outreach. Gail Scowcroft (Graduate School of Oceanogr., Univ. of Rhode Island, 1 Ferry Rd., Narragansett, RI 02882, gailscow@mail.uri.edu) and Kathleen J. Vigness-Raposa (Marine Acoust., Inc., Middletown, RI)

As a complex scientific topic, underwater sound can be challenging for scientists to discuss and effectively communicate with non-science audiences. Educational audiences span formal K-16 classrooms to museum and aquarium visitors. The science of sound is often included in upper middle school physical sciences curricula, high school physics classes, and undergraduate and graduate university courses, which can take advantage of calculus to support student understanding. Communicating with the media presents other challenges: pressing or immediate deadlines; a need to deliver eye-catching, flashy pieces that capture reader attention; and a general lack of fundamental knowledge of underwater sound by readers. Scientists must be proactive in their engagement with media to ensure good

5a FRI. AM

fundamental science is communicated and to increase useful stories about new developments in underwater sound research. Regulators and other decision-makers are also pressed for time when contemplating a topic, yet they need the most up-to-date scientific findings to support their decision-making. This talk will provide an overview of the challenges that ocean acoustic specialists face when trying to communicate the results of their research and meet the needs of diverse audiences. In addition, strategies and possible solutions will be discussed.

9:20

5aAB2. Student engagement and education in underwater sound through the build-a-hydrophone activity. Kevin Joy (Northeast Underwater Res. Technol. and Education Ctr., Univ. of Connecticut, 1080 Shennecossett Rd., Groton, CT 06340, kevin.joy@uconn.edu)

The hydrophone is arguably the most basic sensor used in underwater acoustics. In its simplest form, this passive device provides scientists, and users of all kinds, the ability to listen to and capture natural and anthropogenic sounds that occur in our aquatic world. As part of an array, this technology enables humans to perform such underwater tasks as tracking submerged targets to studying volcanic and tectonic processes. The concept behind the do-it-yourself (DIY) activity to build a hydrophone is not unique. A simple Internet search will reveal numerous design options geared toward constructing a functional hydrophone. The Center for Ocean Sciences Education Excellence (COSEE), Technology and Engineering for Knowledge (TEK), at the University of Connecticut, has modified this activity to offer an alternative that is both affordable and functional, while minimizing the technical skills, tools, and time required for completion. The outcome has been the development of a hands-on student exercise that provides an avenue for engagement and learning in the Science, Technology, Engineering and Mathematics (STEM) fields, while offering a simple means to introduce the science of underwater sound.

9:40

5aAB3. Animations for communicating the science of underwater sound. Daniel A. Russell (Graduate Program in Acoust., Penn State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, drussell@engr.psu.edu)

For several years, the author has been developing a popular website (http://www.acs.psu.edu/drussell/demos.html) consisting of a variety of computer animations illustrating acoustic phenomena involving wave propagation and vibration [Russell, J. Acoust. Soc. Am., **114**, 2308 (2003)]. This paper will showcase a subset of those animations, including several new animations of acoustic wave phenomena and interactive plots created using the Wolfram Mathematica Computable Document Format, which are specifically geared toward enhancing student understanding of the propagation and radiation of sound underwater. Animation examples will include wave propagation; sound radiation from simple sources; directivity patterns for dipoles, doublets, line arrays, and the baffled piston; refraction and reflection; absorption of sound; and waveguides.

10:00

5aAB4. Able sea chicks... Adventures in acoustical oceanography. Lora J. Van Uffelen (Ocean and Resources Eng., Univ. of Hawaii at Manoa, 1000 Pope Rd., MSB 205, Honolulu, HI 96815, loravu@hawaii.edu) and Kathleen E. Wage (Elec. and Comput. Eng., George Mason Univ., Fairfax, VA)

The Able Sea Chicks blog introduces readers to the exciting field of acoustical oceanography by following two female scientists on research cruises to deploy acoustic moorings in the Philippine Sea. The blog was designed for a special outreach session at the April 2010 Acoustical Society of America meeting titled "Listen up and get involved," aimed at girls involved in the Girl Scouts of America. This session included a video explaining some of the principles of underwater sound propagation and showcasing some of the instrumentation used to collect underwater acoustic data. A live, interactive question and answer session with the scientists onboard the research vessel followed the video presentation. Both the blog and the outreach session were designed to reach girls at the age where research has shown that interest in STEM fields begins to decline. The blog illustrates how ocean acoustics experiments are conducted using examples from the PhilSea10 deployment and recovery cruises. Blog posts cover topics such as hydrophone testing, signal processing, mooring deployment, and long-baseline navigation. Since 2010, the Able Sea Chicks video and blog have been used in other Girl Scout sessions at ASA and in talks for high school and university audiences.

10:20

5aAB5. International harmonization of approaches to define underwater noise exposure criteria. Klaus Lucke (CMST, Curtin Univ., GPO Box U1987, Perth, WA 6845, Australia, klaus.lucke@wur.nl), Erwin Winter (Fish Res., IMARES Wageningen UR, IJmuiden, Netherlands), Frans-Peter Lam (Underwater Technol. Dept., TNO, The Hague, Netherlands), Gail Scowcroft (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI), Anthony Hawkins (Loughine Ltd., Aberdeen, United Kingdom), and Arthur N. Popper (Dept. of Biology, Univ. of Maryland, College Park, MD)

An international workshop was held in 2013 with a group comprised of scientists, regulators, and other stakeholders. The workshop focused on how new scientific information related to the effects of underwater noise on marine life influences permitting practices for human activities at sea. Also discussed were how individual countries regulate underwater noise and opportunities for harmonizing approaches on an international scale. The workshop was intended to build momentum toward an international exchange of information and to potentially establish a network for the regulation community. Large gaps in knowledge still exist. In particular, hearing sensitivity in baleen whales, long-term effects of TTS and relevant information on other taxa such as bony fishes, sharks, or invertebrate species, need to be studied more intensively. Regulators need reliable and understandable baseline information on cause-effect relationships. This information could be partially provided through targeted training material for regulators. Another critical regulator need is for opportunities to speak with each other and share knowledge across wide geographic regions. Additional keys to future success are commitments from the regulatory senior management and politicians, invite nations who were not represented in the discussions so far and raise awareness of this topic across a broad audience, including the public.

10:40

5aAB6. Discovery of Sound in the Sea: Resources for regulators, policymakers, and other stakeholders. Kathleen J. Vigness-Raposa (Marine Acoust., Inc., 809 Aquidneck Ave., Middletown, RI 02842, kathleen.vigness@marineacoustics.com), Gail Scowcroft, Christopher Knowlton, and Holly Morin (Graduate School of Oceanogr., Univ. of Rhode Island, Narragansett, RI)

There is concern about the effects of underwater sound on marine life. These effects must be considered by users of underwater sound under several regulations. To fulfill regulatory requirements, up-to-date resources are needed on the potential effects of underwater sound, as well as fundamental science content. The Discovery of Sound in the Sea website (DOSITS; http://www.dosits.org) provides accurate scientific information on underwater sound at levels appropriate for all audiences, including the general public, K-12 teachers and students, college students, policy-makers, and professionals in industry, education, and the media. Content, such as the effects of sound on marine life, is based on peer-reviewed publications and has undergone rigorous review by the DOSITS scientific advisory panel. This talk highlights new resources for regulators, including structured tutorials and a "hot topics" feature. Structured tutorials provide a directed progression of sequential knowledge; the first one outlines the scientific process for determining the risk of exposure to underwater sound, answering the question "How do you determine if a sound source might affect a marine animal?" The "hot topics" feature allows for current, topical issues to be highlighted on the DOSITS front page, providing efficient access to new information, as well as foundational scientific content.

FRIDAY MORNING, 9 MAY 2014 557, 8:45 A.M. TO 12:00 NOON

Session 5aNS

Noise: Aircraft and Fan Noise and Analysis

Nancy S. Timmerman, Chair
Nancy S. Timmerman, P.E., 25 Upton St., Boston, MA 02118

Contributed Papers

8:45

5aNS1. Detection and identification of helicopter noise in beach environments. Jason E. Summers (Appl. Res. in Acoust. LLC, 1222 4th St. SW, Washington, DC 20024-2302, jason.e.summers@ariacoustics.com)

Noise generation by helicopters has been studied extensively as has auditory perception of helicopter noise—both in terms of detection and identification and in terms of annoyance. Recent work [Hoglund *et al*., J. Acoust. Soc. Am. **128**, 164–171 (2010)] has identified the importance of accounting for the specific soundscape when modeling detection performance, suggesting that empirically derived signal-specific detection thresholds must be further adapted to account for differences between real-world ambient environments. Beaches and other coastal environments are an interesting case study for evaluating real-world performance of detection and identification models. In addition to a presenting an unusual and highly dynamic propagation environment for helicopter noise, the natural soundscape resulting from wind and breaking waves has unique spectral and temporal properties that influence how it masks the acoustic signature of helicopters. Here, measurements of a helicopter operating in a beach environment are compared with theoretical and empirical models including models of airborne noise from breaking waves [Bolin and Åbom, J. Acoust. Soc. Am. **127**, 2771–2779 (2010)]. Detection and identification performance are compared with predictions of models for rotorcraft noise—developed in other ambient environments—and models for wind-turbine noise—developed for similar coastal environments.

9:00

5aNS2. Aircraft dose-response research in National Park backcountry areas. Amanda Rapoza, Kristin Lewis, Hastings Aaron, and Cynthia Lee (Volpe National Transportation Systems Ctr., U.S. Dept. of Transportation, 55 Broadway, Cambridge, MA 02142, aaron.hastings@dot.gov)

The Federal Aviation Administration and National Park Service conducted joint research to better understand visitor response to noise from commercial air tour operations over units of the National Park System. Dose-response research in National Parks was conducted during the 1990's at heavily visited, frontcountry sites. To expand this work, aircraft dose-response measurements were recently performed at seven backcountry sites in four National Parks. These sites provided both day and overnight hiking and camping opportunities, with visits ranging from one hour to several days. The sampling approach included side-by-side use of three survey instruments, enabling the evaluation of multiple response variables (annoyance, interference with natural quiet, and acceptability) and comparison of audio clip and *in situ* dose-response, along with the effects of wording and question order. The evaluation of visits longer than previously studied necessitated methodological enhancements including the use of global positioning system based tracking of visitor locations. In total, over 4600 visitor surveys and 50 days of simultaneous acoustic data were collected. This paper, the first of two on this research, describes the research methods and visitor surveys and presents summary results of collected data.

9:15

5aNS3. Aircraft dose-response relations for day-use visitors to backcountry areas in National Parks. Amanda Rapoza (Volpe National Transportation Systems Ctr., U.S. Dept. of Transportation, 55 Broadway, RVT-41, Cambridge, MA 02142, amanda.rapoza@dot.gov), Erika Sudderth (Volpe National Transportation Systems Ctr., Comput. Sci. Corp., Cambridge, MA), Kristin Lewis, Cynthia Lee, and Aaron Hastings (Volpe National Transportation Systems Ctr., U.S. Dept. of Transportation, Cambridge, MA)

The Federal Aviation Administration and National Park Service conducted joint research to better understand visitor response to noise from commercial air tour operations over units of the National Park System. Dose-response relationships developed for heavily visited, frontcountry sites in National Parks showed significant differences in responses between visitors to overlook and short-hike sites, suggesting that activity and visit duration are key influences. An extensive recently collected dataset from backcountry day-use visitors was utilized to further explore these and other influences on dose-response relationships. In this second of two papers on this research, we describe the model-fitting approach used to identify the combination of noise exposure metrics (dose variables) and mediator variables that best predict visitor responses to aircraft noise. The interpretation and application of the best fit model is presented along with the previously developed front-country model. Such dose-response relationships can be used as a tool for evaluating potential impacts of air tours on visitors to National Parks.

9:30

5aNS4. Study on the aerodynamic noise of centrifugal compressors with rotor static eccentricity. Kaijun Wei, Shuguang Zuo, Huijuan He, and Zhe Wang (Clean Energy Automotive Eng. Ctr., Tongji Univ., No. 4800, Cao'an Rd., Jiading District, Shanghai 201804, China, 1110678@tongji.edu.cn)

Centrifugal compressors are widely used in industrial applications. Rotor static eccentricity is among the common faults in centrifugal compressors, and it has a significant influence on the aerodynamic noise. This paper compares the amplitudes and frequency spectrums of the aerodynamic noise of a centrifugal compressor with no eccentricity and the same compressors with different static eccentricities. The aerodynamic noise is calculated by a hybrid method. Large eddy simulation is applied to solve the unsteady three-dimensional internal flow of the compressor, and the pressure fluctuations obtained by the simulation are considered as the noise sources. The far-field aerodynamic noise generated by the noise sources is predicted using the aeroacoustic finite element method. The results show that centrifugal compressors with eccentricity have higher noise level and different spectrum characteristics.

9:45

5aNS5. Interaction noise reduction for a counter-rotating fan by slitted trailing-edge of the forward rotor. Chen Wang and Lixi Huang (Lab of AeroDynam.. and Acoust., HKU Zhejiang Inst. of Res. and Innovation and Dept. of Mech. Eng., The Univ. of Hong Kong, Pokfulam Rd., Hong Kong, Hong Kong, lixi@hku.hk)

A counter-rotating configuration can decrease the size and weight of fans by eliminating stators and increase the efficiency by recovery of the swirl velocity losses. However, that potential is not fully harnessed due to perhaps the issue of noise. This study explores one idea of passive noise reduction for a small axial-flow counter-rotating fan (120 mm in diameter) by the introduction of slitted trailing-edge for the forward rotor. The fan is designed with simple velocity triangle analyses which are checked by 3D steady-flow numerical simulation and experimental measurements of aerodynamic performance. The aerodynamic consequence and the acoustic benefit of such slit geometry are investigated experimentally when the separation distance between the forward and aft rotor is 4 mm. The results show that there is a reduction of total pressure compared with the baseline fan (without slit) at the same rotational speeds, but this is easily compensated for by slightly raising the rotational speeds. A reduction of about 5 dB in overall noise is achieved for the same aerodynamic output in all directions around the fan center. The spectral comparison at the fan inlet indicates that the most prominent interaction noise peaks are suppressed greatly by such trailing edge slits.

10:00–10:15 Break

10:15

5aNS6. Different experimental methods for the measurement of broadband noise sources in ducts. Kunbo Xu and Weiyang Qiao (School of Power and Energy, Northwestern PolyTech. Univ., No.127 Youyi Rd. Beilin District, Xi'an, Shaanxi 710072, China, 364398100@qq.com)

Aeroengine broadband fan noise is a major contributor to the community noise exposure from aircraft. It is currently believed that the dominant broadband noise mechanisms are due to interaction of turbulent wake from the rotor with the stator, and interaction of the turbulent boundary layers on the rotor blades with the trailing edge. Two different methods are presented that enable the separation of different broadband noise sources in turbomachinery ducts, one with respect to modal decomposition developed by DLR, and the other with focused beamformer technique developed in university of Southampton. Different measurement mechanisms are displayed to explain the merits, faults, and requirements.

10:30

5aNS7. Experimental validation of a new intensity estimation method. Benjamin Christensen, Derek C. Thomas, Kent L. Gee, Tracianne B. Neilsen, and Menley Stewart (Brigham Young Univ., 715E 700N, Provo, UT 84606, ukeben@gmail.com)

A new method of estimating acoustic intensity has recently been developed in efforts to improve acoustic measurements of launch vehicles [Thomas *et al.*, "Methods for estimating acoustic intensity in rocket noise fields," JASA **134**(5) 4058–4058 (2013)]. This new method, known as the phase and amplitude gradient estimation (PAGE) method, improves upon the traditional finite difference p-p method for estimating acoustic intensity. The primary advantage of the PAGE method is that it allows for accurate intensity measurements over a larger frequency band. Under certain conditions, it is possible to unwrap the phase component of the PAGE method, allowing for accurate intensity estimates well above previous limitations. The advantages and limitations of the PAGE method are investigated experimentally by taking measurements of arrays of loudspeakers. Preliminary uncertainty analyses of both the PAGE and the finite difference p-p methods are also presented.

10:45

5aNS8. The flow-induced sound of a wall-mounted finite airfoil. Danielle Moreau and Con J. Doolan (School of Mech. Eng., Univ. of Adelaide, North Terrace, Adelaide, SA 5005, Australia, danielle.moreau@adelaide.edu.au)

In this study, the aeroacoustic behavior of a wall-mounted finite airfoil is experimentally investigated. Compared to a semi-infinite or two-dimensional airfoil where end effects are not considered, a wall-mounted finite airfoil is more realistic, especially for applications such as wind turbine blades attached to a hub, submarine hydrofoils mounted to a hull, or stators connected to a hub or outer wall. Acoustic and aerodynamic measurements have been taken in an anechoic wind tunnel with single microphones, multiple acoustic beamforming arrays, and hot-wire anemometry. These measurements are used to examine changes in flow topology and radiated noise as a function of Reynolds number and airfoil aspect ratio. Additionally, the data gives insight into the influence of flow at the airfoil tip and wall junction on noise production.

11:00

5aNS9. An experimental research on the wake flow structure of serrated cascade. Kunbo Xu and Qiao Weiyang (School of Power and Energy, Northwestern PolyTech. Univ., No.127 Youyi Rd., Beilin District, Xi'an, Shaanxi 710072, China, 364398100@qq.com)

This study concerns the mechanisms of the turbulent broadband noise reduction for cascade trailing edge serrations while the inlet velocity changed from 30 m/s to 50 m/s. The turbulence spatio-temporal information are measured with 3D hot-wire. The experiment is carried out in the Northwestern Polytechnical University low speed open jet wind tunnel. It showed the spreading rate of the wake and the decay rate of the wake centerline velocity deficit increased with serrated edge compared to the straight edge, and the three components of velocity changed differently with serrated trailing edge. It is also found that the turbulence peak occurs further from the airfoil surface in the presence of the serrations, and the serrations widened the mix area, which allowed the flow mixed together ahead of the schedule.

11:15

5aNS10. An equivalent source model for the sound intensity in the vicinity of a high-performance military aircraft. Trevor A. Stout, Gee L. Kent, Tracianne B. Neilsen, Benjamin Y. Christensen, Derek C. Thomas (Phys., Brigham Young Univ., 688 North 500 East, Provo, UT 84606, titorep@gmail.com), and Michael M. James (Blue Ridge Res. and Consulting LLC, Asheville, NC)

The sound field of the F-22A Raptor has been measured extensively using arrays of microphones including an intensity probe to the sideline and rear of the aircraft. Recently, an equivalent source model (ESM) has been proposed that incorporates two arrays of monopole sources placed along the jet axis and their image sources to account for the hard ground. The

monopole amplitudes in each case are described by a Rayleigh distribution, and one array includes a phase angle between the monopoles to produce correlated noise steered in a specific direction. The model closely replicates the measured SPL at the majority of measurement locations. However, agreement with energy-based quantities such as the vector acoustic intensity may not follow from agreement with the pressure measurements alone. Hence, in this work, the ESM's ability to produce intensity estimates that match the measured F-22A intensity is evaluated. A new method of estimating acoustic intensity, known as the phase and amplitude gradient estimation (PAGE) method, is applied to the measured F-22A data. The PAGE method accurately estimates intensity in a larger bandwidth than the traditional finite difference method, expanding this discussion of the ESM's accuracy. In addition, sensitivity of ESM parameter selection, e.g., correlated source phase angle and peak amplitude location, on changes in the predicted intensity is examined. [Work sponsored by ONR.]

11:30

5aNS11. Military noise limits: Design criteria versus operational effectiveness. Bruce E. Amrein (Human Res. & Eng. Directorate, Army Res. Lab., Perceptual Sci. Branch, 520 Mulberry Point Rd., Aberdeen Proving Ground, MD 21005, bruce.e.amrein.civ@mail.mil) and Tomasz R. Letowski (Human Res. & Eng. Directorate, Army Res. Lab., Aberedeen Proving Ground, MD)

Noise requirements in military environments differ significantly from typical industrial or occupational situations. Both in combat and in training, mission success requires offensive equipment and weapons to be more lethal and survivable than those used by the adversary. Higher muzzle velocities, heavier projectiles, and more powerful engines result in high levels of both steady-state and impulsive noise and an increased risk of hearing loss for the users. Weapons firing can expose the user to more energy in a single event than typically experienced in a working lifetime of occupational exposure. In addition, military operations require effective speech communication while minimizing auditory detection of equipment by the adversary. Producing material suitable for various forms of military operations requires unique design criteria often exceeding civilian national or international standards. To meet these unique and often contradicting requirements, the U.S. military developed a military design standard for noise limits. This standard (MIL-STD-1474) was last revised in 1997. This paper describes the effort of the U.S. Army, Navy, and Air Force to update the standard to permit production and fielding of military systems designed to maximize Warfighter effectiveness, while minimizing hearing damage caused by their use.

11:45

5aNS12. Development of a digital noise exposure system for research on noise induced hearing loss. Jun Qin (Elec. and Comput. Eng., Southern Illinois Univ. Carbondale, 1230 Lincoln Dr., ENGR-E207, Mail Code: 6603, Carbondale, IL 62901, jqin@siu.edu)

Over 30 million Americans suffer from noise induced hearing loss (NIHL). Many research projects demonstrated that different types of noise, even with equal sound energies, could produce different amounts of hearing loss. In this project, a novel digital noise exposure system has been developed for generating various noise signals (e.g., pure-tone, Gaussian, impulsive, and complex noise). The developed system can be used to study NIHL produced by different types of noise in an animal model. The system could produce impulse noise with peak sound pressure level (SPL) up to 160 dB, which effectively mimics the noise generated by a military weapon (e.g., M-16 rifle). In addition, continuous Gaussian noise with peak SPL up to 140 dB can be created, which is well above the 85 dB recommended exposure limit established by the National Institute for Occupational Safety and Health (NIOSH). The preliminary results of in an animal study showed significant permanent threshold shift (PTS) produced by 90 shocks impulse noise with peak SPL = 155 dB generated by the system. In summary, the digital noise exposure system replicates environmental noise allowing researchers to study hearing loss in a controlled situation.

Session 5aPA

Physical Acoustics: General Topics in Physical Acoustics

David A. Brown, Chair
ATMC/ECE, Univ. of Massachusetts Dartmouth, 151 Martine St., Fall River, MA 02723

Contributed Papers

9:00

5aPA1. Trillium: A thermoacoustic-Stirling refrigerator implementation using an inline topology. Matthew E. Poese, Robert M. Keolian, Robert W. Smith, Eric C. Mitchell, Cory M. Roberts, and Steven L. Garrett (Appl. Res. Lab., The Penn State Univ., PO Box 30, State College, PA 16804, keolian@psu.edu)

To meet a small but growing demand for low global-warming potential refrigeration technologies, a thermoacoustic-Stirling refrigerator has been developed that uses helium as the working gas. Using a new multi-stage topology [Backhaus and Keolian, "In-line stirling energy system," US patent 7,908,856 B2 (2011)], Trillium employs a collinear stack of three linear motors driven by voltage signals phased 120 degrees apart from one another. Above each motor is a thermal core that is comprised of a pair of novel aluminum microchannel heat exchangers on either side of a rolled Kapton® regenerator. The heat exchangers have an equivalent pore size of about a thermal boundary layer thicknesses that allows high effectiveness. The linear motors drive an acoustic wave that travels along the length of the machine. Taken in isolation, each thermal core and the motor-driven piston above and below (sealed using novel plastic flexure seals) forms a single alpha-Stirling machine; taken together Trillium is three Stirling machines working together with, in principle, zero vibration due to a stationary center-of-mass as a result of the motor phasing. The talk will explain the design and present performance measurements. [Work generously sponsored the Advanced Research Project Agency—Energy (ARPA-E) under their BEETIT program.]

9:15

5aPA2. Gedeon streaming suppression in a small scale thermoacoustic-Stirling engine-generator. D. Wilcox and P. Spoor (Chart Inc. - QDr., 302 Tenth St., Troy, NY 12180, douglas.wilcox@chartindustries.com)

Thermoacoustic-Stirling engines, or traveling wave engines, have been shown to convert heat to acoustic power very efficiently (over 30% first-law) in the laboratory. The laboratory prototypes are generally heated from the inside by an embedded electric heater, have a long, bulky resonator, and deliver their work to an acoustic load rather than as electricity, leaving significant challenges required for commercialization unaddressed. The authors are part of a team developing a compact acoustic Stirling engine that is externally heated and is coupled to a pair of linear alternators, dubbed the Thermoacoustic-Stirling Engine-Generator (TaSEG). An important part of this work has been developing a commercially viable means of suppressing Gedeon streaming, a steady flow that circulates in an acoustic engine's toroidal geometry. In the laboratory, this streaming is typically suppressed by either a latex barrier or a "jet pump,'' a special flow element with asymmetric flow resistance, adjusted from the outside of the engine via a rod that passes through the pressure vessel. This work describes the design and testing of a simple, compact, and inexpensive element with multiple jet-pump orifices (the "jet plate''), which can replace the laboratory versions in a commercial engine.

9:30

5aPA3. Bayesian-based model selection and physical parameter estimation of the acoustical properties of rigid-frame porous media. Cameron J. Fackler, Ning Xiang (Graduate Program in Architectural Acoust., Rensselaer Polytechnic Inst., 110 8th St, Greene Bldg., Troy, NY 12180, facklc@rpi.edu), Kirill V. Horoshenkov (Dept. of Mech. Eng., Univ. of Sheffield, Sheffield, United Kingdom), and Amir Khan (School of Eng. and Informatics, Univ. of Bradford, Bradford, United Kingdom)

This paper studies an application of Bayesian inference to determine the most appropriate model for the acoustical properties of rigid-frame porous media. The Bayesian framework encompasses two levels of inference: model selection and parameter estimation. Both are based on the posterior probability distribution, which is unique to each combination of acoustic model and material sample. The posterior distribution provides statistically informed guidance on selecting an appropriate acoustic model for a material sample as well as determining the physical parameters of the porous microstructure. Using the nested sampling algorithm, the posterior distribution for each material sample and acoustic model is explored and integrated numerically. This process computes the Bayesian evidence for each model, which is used to select a suitable model for the acoustical properties of the material. The process also provides estimates of the physical parameters, which are used with the selected model for accurate prediction of the acoustical properties of the material.

9:45

5aPA4. Modeling of sound propagation in the vicinity of rigid porous ground by boundary element method. Yiming Wang and Kai Ming Li (Mech. Eng., Ray W. Herrick Labs., Purdue Univ., 140 South Martin Jischke Dr., West Lafayette, IN 47907-2031, wang1679@purdue.edu)

A standard boundary integral equation (BIE) technique takes advantage of the well-known Green's function for the sound fields above an impedance ground. It leads to a simplified solution that enables the development of efficient two-dimensional boundary element numerical codes for computing sound fields above a locally reacting ground. The method is particularly useful if there are multiple reflecting surfaces and the presence of a mixed impedance ground in the proximity of sources and receivers. However, the BIE formulation cannot be applied readily for calculating the sound fields above a non-locally reacting ground because the corresponding Green's function is generally not available. The so-called two-domain approach was frequently used to model the sound propagation inside the porous material by assuming it as a dissipative fluid medium. By matching the particle velocities and pressures at the interface, a two-domain BIE formulation can be developed for modeling the sound propagation above a rigid porous ground. Use of the two-domain approach has a significant impact on the required computational resources. The current paper presents an alternative method aiming to reduce the computational time by exploring the use of an accurate Green's function for the development of a BIE formulation above a non-locally reacting ground.

10:00

5aPA5. Convergence behavior and steady state response of a rib-stiffened, layered plate structure subjected to high frequency acoustic loading. Jeffrey Cipolla (Weidlinger Assoc., Inc., 1825 K St. NW, Ste. 350, Washington, DC 20006, cipolla@wai.com), Patrick Murray, and Kirubel Teferra (Weidlinger Assoc., Inc., New York, Uganda)

An existing analytical, frequency domain solution for wave propagation in coated, ribbed, three-dimensional elastic layered plates excited by acoustic plane waves provides fast solutions for high frequency excitations. The solution methodology, which is found to be numerically unstable under certain conditions, contains an Ansatz for a particular wave number expansion in the direction of periodicity. Evidence is presented to show that the numerical instability is due to the specific choice of the wave number basis. In order to provide a remedy while retaining the positive aspects of the solution methodology, we determine the set of admissible propagating (and attenuating) waves via an eigenvalue analysis. Several approaches exist to determine the admissible waves of structures with periodicity. The Wave Guide Finite Element (WFE) method leads to a two parameter, nonlinear eigenvalue problem, which is difficult to solve. The Scale Independent Element (SIE) formulation results in a two-parameter quadratic eigenvalue problem and overcomes the numerical issues of the WFE method. This study examines available methods in determining the admissible waves and compares them with those established using the dispersion relation. The computed admissible waves are then compared with the aforementioned Ansatz.

10:15–10:45 Break

10:45

5aPA6. An iterative approach to measurement of oblique acoustic absorption coefficient in three-dimensions. Hubert S. Hall (Naval Surface Warfare Ctr. Carderock, 620 Michigan Ave. NE, Washington, District of Columbia 20064, 61hall@cardinalmail.cua.edu), Joseph F. Vignola, John Judge (Dept. of Mech. Eng., The Catholic Univ. of America, Washington, DC), and Diego Turo (Dept. of BioEng., George Mason Univ., Fairfax, VA)

The measurement of oblique acoustic absorption coefficient remains an ill-defined test. Unlike Kundt's tube (ASTM C384 and E1050) and Sabine room measurements (ASTM C423), there is no uniform test method or standard to follow. *In-situ* or specific application based testing is the industry norm for arbitrary angular absorption coefficient measurement. An iterative approach has been developed for a free/free/free/free panel in an anechoic chamber. Using numerical model predictions of the acoustic field, microphone measurement data are updated to account for a multitude of acoustic phenomena including diffraction from the panel. This method provides a quick measurement option that is flexible and transportable for many applications.

11:00

5aPA7. Infrasound wind noise reduction via porous fabric domes. John M. Noble, W. C. Kirkpatrick Alberts (US Army Reserach Lab., 2800 Powder Mill, Adelphi, MD 20783, kirkalberts@verizon.net), Richard Raspet (National Ctr. for Physical Acoust., Univ. of MS, University, MS), Sandra L. Collier, and Mark A. Coleman (US Army Reserach Lab., Adelphi, MD)

Often, porous hose arrays or complex pipe rosettes are used for wind noise reduction in long-term infrasound monitoring situations. Both of these wind noise reduction schemes perform well in reducing the wind-generated noise at an infrasound sensor, but each can significantly filter the information reaching the sensor. Ideally, a wind noise reduction scheme should have a flat frequency response and a minimal attenuation over the desired frequency band. To that end, three fabrics, two porous and one non-porous, stretched in dome configurations have been investigated for wind noise reduction. Two porous fabric domes are found to perform comparably to a porous hose array but without its amplitude- and phase-altering properties.

11:15

5aPA8. Focused ultrasonic pulsed source for non-contact measurements in air. Frederic Cohen Tenoudji, Dominique Busquet, Jean François Mourey (Institut Jean le Rond d Alembert,MPIA, Sorbonne Universités, UPMC Univ. Paris 06, CNRS, UMR 7190, 2, Pl. de la Gare de Ceinture, Saint-Cyr-l 78210, France, fcohentenoudji@yahoo.fr),

Ultrasonic spark sources have qualities propitious to non-destructive testing of materials in air. They are energetic, and they generate a short acoustic pulse which bandwidth extends from a few hundred Hertz to several hundred kilohertz. It is shown here that by focusing the cylindrically symmetric wavefront generated by the spark with an elliptical mirror, it is possible to concentrate the acoustic energy on the focal line image of the spark. This technique provides a powerful virtual source of small dimension, which may be localized on the surface of the material to be inspected. The performance of this source for non-contact inspection in transmission of composite materials is evaluated by measurements on test samples. The lateral spatial resolution and the high frequency limitation of this technique are determined.

11:30

5aPA9. Experimental research on a sound insulation structure based on embedded Helmholtz resonators. Xinwu Zeng, Dongbao Gao, and Changchao Gong (College of OptoElectron. Sci. and Eng., National Univ. of Defense Technol., Yanzheng St., Changsha 410073, China, xinwuzeng@nudt.edu.cn)

The structures containing Helmholtz resonators (HRs) have been demonstrated to be one of the metamaterials with negative parameters. Due to its local resonance, the one- and two-dimensional structures were indicated having local resonant forbidden band, which are possible ways to attenuate low-frequency acoustic waves and oscillations. At the resonant frequency, the incident acoustic energy is almost localized around the resonator, when a local resonant gap (LRG) appears at the end of the structure. Since the existence of LRG is only associated with the properties of single resonator, a wide band gap can be produced using periodically arrayed HRs with gradually changed parameters. In this structure, neighbor forbidden gaps are overlapped. A sound insulation device is designed and implemented in this paper based on embedded Helmholtz resonators (HRs). This structure is feasible to insulate acoustic waves transmitting above it. Numerical and experimental results both show that a low-frequency wideband insulation area can be formed by circularly arrayed HRs. The working band can be adjusted by changing the resonant frequency of the HRs in each layer.

11:45

5aPA10. Energy flux streamlines versus acoustic rays for modeling an acoustic lens: Energy flow inside and in the focal region for a carbon dioxide filled spherical balloon in air. 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 [Dean and 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)], a comparison of ray acoustics and wave analysis via energy flux streamlines [Chapman, "Using streamlines to visualize acoustic energy flow across boundaries," J. Acoust. Soc. Am. **124**, 48–56 (2008)] as a means of visualizing the sound field for a positive acoustic lens in the form of a carbon dioxide filled spherical balloon in air is made. The sound field is expanded in the usual Legendre polynomials and spherical Bessel functions [Anderson, "Sound scattering from a fluid sphere," J. Acoust. Soc. Am. **22**, 426–431 (1950)], and the energy flux vectors at points throughout the regions of interest are calculated [Adin Mann III, *et al*., "Instantaneous and time-averaged energy transfer in acoustic fields," J. Acoust. Soc. Am. **82**, 17–30 (1987)]. Then, energy flux streamlines are plotted using Mathematica routines for comparison with conventional acoustical rays, both inside and outside the balloon. Particular attention is paid to the focal region.

Session 5aPP

Psychological and Physiological Acoustics: Potpourri (Poster Session)

Samuel R. Mathias, Chair
Ctr. for Computational Neuroscience and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors an opportunity to see other posters, contributors of odd-numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and contributors of even-numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

5aPP1. A "better ear" listening strategy for improving speech-in-noise understanding in bilateral cochlear implant users. Alan Kan and Ruth Y. Litovsky (Univ. of Wisconsin-Madison, 1500 Highland Ave., Madison, WI 53705, ahkan@waisman.wisc.edu)

For bilateral cochlear implant (BiCI) users, understanding a target talker in noisy situations is difficult. Current efforts for improving speech-in-noise understanding have focused on improving signal-to-noise ratio (SNR) by multi-microphone techniques and signal processing, with only moderate improvement in performance. BiCI users typically report having a "better ear" for listening and recent data collected in our lab has shown that they have an asymmetry in speech unmasking performance. This work proposes a novel listening strategy for improving speech-in-noise understanding by combining: (1) a priori knowledge of a "better ear" and having a BiCI user selectively attend to a target talker in that ear; with (2) signal processing that delivers the target talker to the "better" ear and the noisy background to the opposite ear. We compared performance on a speech-in-noise test with and without this "better ear" strategy, using a virtual auditory space created from individualized head-related transfer functions. Subjects showed an improvement of at least 6 dB SNR in the speech reception threshold when using the "better ear" strategy, demonstrating that the strategy can boost speech-in-noise understanding for BiCI users. The novelty of this strategy is that it can be easily applied to other devices aimed at improving hearing.

5aPP2. Preliminary evaluation of a physiologically inspired signal processing strategy for cochlear implants. Jayaganesh Swaminathan (Sensimetrics Corp., 635 Commonwealth Ave., Rm. 320, Boston, MA 02215, jswamy@bu.edu), Raymond L. Goldsworthy, Patrick M. Zurek (Sensimetrics Corp., Malden, MA), Agnès C. Léger, and Louis D. Braida (Res. Lab. of Electronics, MIT, Cambridge, MA)

An approach to developing a cochlear-implant (CI) signal processing strategy inspired by healthy auditory physiology is described, and preliminary perceptual measurements with CI patients are reported. Envelopes were derived from a phenomenological auditory-nerve model (Zilany and Carney, 2010) that has been rigorously tested against physiological responses to both simple and complex stimuli. The model captures several physiological properties associated with nonlinear cochlear tuning including compression, suppression, and neural adaptation that are known to be critical for the representation of speech in quiet and noise. The envelopes were imposed on constant-rate current pulses via a programmable interface. Performance with this "neural" strategy was compared against the listeners' personal processor and a standard encoder (ACE) processor implementation. Sentence, consonant and vowel identification were obtained from 5 CI users in quiet and +6 dB of continuous speech-shaped noise. Overall, performance with the neural strategy was intermediate between that with subjects' personal processors and the standard ACE processor. The closeness of results with the neural strategy to the listeners' personal processor was viewed as promising as no individual tailoring of parameters were performed, and the subjects had no training with the neural strategy. Results from modeling efforts and future research directions will be discussed. [Work supported by NIH-NIDCD (R43-DC013006).]

5aPP3. Speech understanding in adults and children for sentences with cochlear-implant simulated spectral degradation and shallow insertion depth. Sara Dougherty, Arifi Waked, and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland, 0119E Lefrak Hall, College Park, College Park, MD 20742, saraclaire216@gmail.com)

While previous studies have compared cochlear-implant simulated (i.e., vocoded) speech understanding between adults and children, they have been acute experiments that have not considered long-term adaptation or training effects. Normal-hearing adult listeners demonstrate significant improvement in vocoded speech understanding with training, particularly if there is simulated frequency-to-place mismatch. The purpose of this study was to compare how adults and children adapt to eight-channel sine-vocoded speech with 0, 3, and 6 mm of frequency-to-place mismatch. Twenty adults (>18 yrs) and ten children (8–10 yrs) were trained on vocoded speech understanding over a four-hour period. The stimuli were a closed set of 40 simple words from which five-word nonsense sentences were constructed. Speech understanding was measured in 45-trial blocks where no feedback was provided, followed by 30-trial blocks where visual and auditory feedback was provided. High variability existed within both groups. On average, children performed worse than adults. Over the first five testing/training blocks, children improved at a slower rate than adults. Some children showed minimal improvement over the testing, whereas most of the adults showed noticeable improvement. These results suggest that adults have developed neural mechanisms that can more effectively adapt to and process degraded and frequency-shifted speech.

5aPP4. Discrimination of inconsistent interaural time differences across frequency in simulated bilateral cochlear-implant users. Francisco A. Rodríguez and Matthew J. Goupell (Hearing and Speech Sci., Univ. of Maryland, 0119E Lefrak Hall, College Park, MD 20742, frodri@gmail.com)

When using highly controlled low-rate stimulation, interaural-time-difference (ITD) just-noticeable differences (JNDs) vary greatly across frequency and individual bilateral cochlear-implant (CI) users. It is unclear how across-frequency variability would affect JNDs in multi-electrode stimulation where there are channel interactions from current spread. Therefore, we simulated variable across-frequency ITD sensitivity and channel interactions in 10 normal-hearing listeners presented with a CI simulation that consisted of 100-pulse/s, 1.5-mm bandlimited pulse trains (PTs) in three frequency regions (middle frequency = 7.6 kHz). The simulated electrode spacing was varied by placing the low- and high-frequency PTs $\Delta = 0.75$, 1.5, 3, or 4.5 mm from the middle frequency. Inconsistent ITD sensitivity across frequency was simulated by applying non-zero ITDs in one, two, or three PTs, while the remaining PTs had zero ITD. The pulses in the PTs were simultaneous, nearly simultaneous, or maximally separated. In a left-right discrimination task, JNDs increased for a decreasing number of dichotic PTs. JNDs increased for $\Delta < 3$ mm, suggesting an important role of channel interactions. JNDs increased for non-simultaneous stimulation likely due to a rate limitation effect. The relatively poorer listener performance from the factors in our simulations suggest a means to predict multi-electrode ITD performance in CI users.

5aPP5. Auditory evoked responses to perceived quality of vocoded speech. Chin-Tuan Tan, Elizabeth Glassman, Samuel Oh (Dept. of Otolaryngol., New York Univ., School of Medicine, 550 First Ave. NBV 5E5, NY 10016, Chin-Tuan.Tan@nyumc.org), WenJie Wang, Brett Martin (Speech and Hearing Program, City Univ. of New York, Graduate Ctr., New York, NY), Arlene Neuman, and Mario Svirsky (Dept. of Otolaryngol., New York Univ., School of Medicine, New York, NY)

One of the common uses of vocoded speech is to simulate what speech would sound like when it is processed via a cochlear implant. However, listeners with normal hearing may perceive vocoded speech differently depending on the carrier signals (noise or tone) or channel bandwidths. In this study, we attempted to determine how physiological measures correlate with the quality ratings of vocoded speech that varied in carrier signal and channel bandwidth. Eleven NH subjects rated the perceived sound quality of a spoken sentence processed with noise and tone vocoders (22-channel analysis/synthesis filterbank) whose channels were broadened and narrowed by different Q factors. Each subject was instructed to rate sound quality on a 10 point scale where "10" indicates "clean" and "1" represents "very-distorted". Quality ratings for six different combinations of carrier and Q factor were very consistent across listeners. Auditory evoked potentials (MMN) were obtained from three NH listeners. Vowel stimuli processed through the vocoders with variations in the Q value were used as stimuli. Changes in N1 latency were observed to correlate with the perceived quality ratings of the vocoded speech. [This study was supported by NIH-K25-DC010834 (PI: Tan), PSC-CUNY (PI: Martin), NIH-R01-DC011329 (PI: Neuman and Svirsky), and NIH-R01-DC003937 (PI: Svirsky).]

5aPP6. The precedence effect: Exploring the build-up of echo thresholds in cochlear-implant listeners. Tanvi D. Thakkar (Commun. Sci. and Disord., Univ. of Wisconsin-Madison, 1500 Highland Ave., Rm. 563, Madison, WI 53705, tthakkar@wisc.edu), Andrew D. Brown (Dept. of Physiol. & Biophys., Univ. of Colorado SOM, Aurora, CO), Heath G. Jones, Alan Kan, and Ruth Y. Litovsky (Binaural Hearing & Speech Lab, Waisman Ctr., Univ. of Wisconsin-Madison, Madison, WI)

In reverberant environments, listeners rely on early arriving spatial cues to accurately localize sound sources, a phenomenon known as the precedence effect (PE). In deaf individuals fitted with bilateral cochlear implants (BiCIs), this effect is diminished, ostensibly due to the fact that clinical processers do not preserve binaural cues. We have recently demonstrated that BiCI listeners do exhibit aspects of the PE similar to normal-hearing (NH) listeners when binaural stimulation is restored using synchronized research devices. Here, we consider whether BiCI users also demonstrate an aspect of the PE known as "buildup"—enhancement of the PE after a repeated stimulus. BiCI users with demonstrated sensitivity to interaural time differences (ITDs) were tested using dichotic electrical pulses ($\pm$500μs ITD in opposing "lead" and "lag" pulse pairs, with lead-lag delays of 1–64 ms). On each trial, listeners indicated (1) whether one or two locations were perceived (to assess "fusion") and (2) the location perceived (or, given two locations, the "left-most" location perceived, to assess "localization dominance"). Preliminary results indicate that 'buildup' may be atypical in BiCI users, who have experienced years of acoustic deprivation. Lack of adaptation to redundant stimuli may temper the extent of benefit from restored binaural inputs in reverberant environments.

5aPP7. Recovery from forward masking of vowels and consonants: Effects of age and hearing loss. William J. Bologna (Dept. of Hearing and Speech Sci., Univ. of Maryland, 135 Rutledge Ave, MSC 550, Charleston, SC 29425, bologna@musc.edu), Daniel Fogerty (Dept. of Commun. Sci. and Disord., Univ. of South Carolina, Columbia, SC), Jayne B. Ahlstrom, and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

Vowels contain slow periodic fluctuations that contrast with aperiodic noise, which may favor perception of vowels over consonants with simultaneous and forward noise maskers. Older adults with or without hearing loss may be poorer at coding these temporal periodicity cues, which may limit their performance. In this study, younger adults with normal hearing and older adults with normal and impaired hearing identified consonants or vowels in the initial or final position of noise-masked syllables. Syllable and masker duration and relative timing ensured that the final 10, 40, or 100 ms of the syllable occurred after masker offset. Individualized spectral shaping minimized confounding effects of reduced audibility in regions of hearing loss. Preliminary results for initial-position phonemes indicate vowels are less susceptible to simultaneous masking than consonants. Recognition of final-position vowels is facilitated by 40-ms delay, whereas final consonants require 100-ms delay or longer for similar improvement. Younger adults benefit most from these delays, older adults with normal hearing benefit less, and older adults with hearing loss benefit least. These findings suggest that age and hearing loss contribute to prolonged recovery from forward masking, and that vowels have greater resistance to forward masking than consonants. [Work supported by NIH/NIDCD and ASHA.]

5aPP8. Modeling individual differences in overshoot: Effects of age, hearing loss, and efferent feedback. Skyler G. Jennings (Commun. Sci. and Disord., Univ. of Utah, 390 S. 1530 E., Salt Lake City, UT 84112, skyler.jennings@hsc.utah.edu), Jayne B. Ahlstrom, and Judy R. Dubno (Dept. of Otolaryngology-Head and Neck Surgery, Medical Univ. of South Carolina, Charleston, SC)

The detection of a short sinusoidal probe in simultaneous masking improves as the probe's onset is delayed from the masker's onset. This "overshoot" may be mediated by the medial olivocochlear (MOC) reflex, whose pathway includes spiral ganglion neurons (SGNs), olivocochlear neurons and the outer hair cells (OHCs). Overshoot was measured in younger adults with normal hearing, older adults with normal hearing, and older adults with hearing loss to test the hypothesis that overshoot decreases as components in the MOC reflex pathway are compromised. Overshoot was significantly reduced in older adults, but only those with hearing loss, which is consistent with overshoot depending primarily on the status of the OHCs and only minimally influenced by age-related reductions in SGNs. Thresholds measured when the probe was near the masker's onset showed large differences across listeners, resulting in appreciable individual differences in overshoot. Simulations were generated from a computational model of the auditory system to quantify the contributions of cochlear hearing loss, MOC reflex strength, and detection efficiency to individual differences in overshoot. Preliminary results suggest that cochlear hearing loss and detection efficiency explain the largest portion of the variance in overshoot among adults with normal and impaired hearing. [Work supported by NIH/NIDCD.]

5aPP9. Modulation masking attributes of narrowband and low-noise noise forward maskers in normal-hearing and hearing-impaired listeners. Adam Svec, Peggy B. Nelson (Dept. of Speech-Language-Hearing Sci., Univ. of Minnesota,Minneapolis, MN 55455, svecx002@umn.edu), and Judy Dubno (Dept. of Otolaryngol., Medical Univ. of South Carolina, Charleston, SC)

Savel and Bacon (2003) measured detection thresholds for a 4000 Hz pure-tone signal in the presence of a narrowband noise (NBN) or a low-noise noise (LNN) simultaneous masker. The authors asserted that fluctuations in the envelope of the NBN were likely responsible for its increased masking effectiveness. Because modulation detection interference (MDI) is larger for hearing-impaired (HI) than normal-hearing (NH) listeners using modulated simultaneous maskers (e.g., Lorenzi *et al.*, 1997) and forward maskers (e.g., Koopman *et al.*, 2008), we measured detection thresholds for NH and HI listeners for pure tones in the presence of either NBN or LNN forward maskers that were either 1-ERB wide or 1/3-ERB wide. Based on previous MDI findings (Koopman *et al.*, 2008), we predicted larger differences in masked thresholds for the NBN and LNN conditions for HI than NH listeners. These results for detection thresholds for pure-tone signals have implications for interpreting differences in modulated forward masking for NH and HI listeners. [Work supported by grants from NIH/NIDCD.]

5aPP10. Neurometric amplitude-modulation detection threshold measured in the chinchilla ventral cochlear nucleus following sensorineural hearing loss. Mark Sayles, Ann E. Hickox (Dept. of Speech, Lang. and Hearing Sci., Purdue Univ., Heavilon Hall, West Lafayette, IN 47907, sayles.m@gmail.com), and Michael G. Heinz (Dept. of Biomedical Eng., Purdue Univ., West Lafayette, IN)

Amplitude modulation is a common feature of natural sounds and an important cue in audition. Modulation supports perceptual segregation of "objects" in complex acoustic scenes, and provides information for speech understanding and pitch perception. Previous work in our laboratory showed increased modulation gain without change in temporal modulation transfer function (tMTF) bandwidth in auditory-nerve fiber responses to sinusoidal amplitude-modulated (SAM) tones measured in chinchillas with noise-induced hearing loss (HL), compared to normal-hearing (NH) controls. The ventral cochlear nucleus (VCN) provides significant input-output transformations with respect to amplitude-modulation representation, with enhanced spike synchrony to the amplitude envelope in several distinct cell types. We recorded spike times in response to SAM tones with modulation depths between 3% and 100% from all major VCN unit types in anesthetized NH and HL chinchillas. HL animals were previously exposed to 116 dBSPL 500 Hz-centered octave-band Gaussian noise. Spike times were analyzed in terms of synchrony to the amplitude envelope, tMTFs were calculated, and a signal-detection theoretic analysis was used to compute modulation-detection and discrimination thresholds. Results will be related to human perceptual studies, which have shown better modulation-detection thresholds in HL. [Work supported by an Action on Hearing Loss Fulbright Commission scholarship (M.S.), and NIH grant R01-DC009838.]

5aPP11. An algorithm to improve speech recognition in noise for hearing-impaired listeners: Consonant identification and articulatory feature transmission. Eric W. Healy, Sarah E. Yoho (Speech & Hearing Sci., The Ohio State Univ., 1070 Carmack Rd., Columbus, OH 43017, healy.66@osu.edu), Yuxuan Wang (Comput. Sci. & Eng., The Ohio State Univ., Columbus, OH), Frederic Apoux, Carla L. Youngdahl (Speech & Hearing Sci., The Ohio State Univ., Columbus, OH), and DeLiang Wang (Comput. Sci. & Eng., The Ohio State Univ., Columbus, OH)

Previous work has shown that a supervised-learning algorithm estimating the ideal binary mask (IBM) can improve sentence intelligibility in noise for hearing-impaired (HI) listeners from scores below 30% to above 80% [Healy *et al.*, J. Acoust. Soc. Am. **134** (2013)]. The algorithm generates a binary mask by using a deep neural network to classify speech-dominant and noise-dominant time-frequency units. In the current study, these results are extended to consonant recognition, in order to examine the specific speech cues responsible for the observed performance improvements. Consonant recognition in speech-shaped noise or babble was examined in normal-hearing and HI listeners in three conditions: unprocessed, noise removed via the IBM, and noise removed via the classification-based algorithm. The IBM demonstrated substantial performance improvements, averaging up to 45% points. The algorithm also produced sizeable gains, averaging up to 34% points. An information-transmission analysis of cues associated with manner of articulation, place of articulation, and voicing indicated general similarity in the cues transmitted by the IBM versus the algorithm. However, important differences were observed, which may guide the future refinement of the algorithm. [Work supported by NIH.]

5aPP12. Optimizing masker phase effects for use in a portable hearing screening tool. Evelyn M. Hoglund, Lawrence L. Feth, Yonghee Oh, and Niall Klyn (Speech and Hearing Sci., The Ohio State Univ., 110 Pressey Hall, 1070 Carmack Rd., Columbus, OH 43210, hoglund.1@osu.edu)

The current study is a continuation of work toward development of a preclinical indicator for noise induced hearing loss. Masked threshold differences produced by Schroeder-phase maskers have been demonstrated with long duration tones at different frequencies (1, 2, and 4 kHz) as well as with IEEE sentences (Summers and Leek, 1998) for normal hearing listeners, but these differences are not found for hearing impaired listeners. Similar results are also found for repeated short duration tone bursts and greater differences occur with systematically changed tone burst frequencies (Hoglund *et al.*, 2012). Using the enhanced channel model (Oh, 2013), threshold differences are predicted, and the Schroeder-phase masker characteristics are adjusted to maximize the difference. This leads to a greater range of masker phase effect differences between the positive and negative Schroeder-phase maskers, and should allow greater sensitivity to preclinical hearing threshold changes. These optimized maskers were also applied to a digit triplets test designed for telephone hearing screening (Watson *et al.*, 2012). Testing with both the digit triplets and the short duration tone bursts are compared for ease of use and portability, as well as sensitivity to small changes in post-exposure thresholds. [Work supported by the Office of Naval Research.]

5aPP13. Acceptable noise level in bilinguals: Consideration of signal and masker effects. Gabrielly Azcona (Commun. Sci. and Disord., Long Island Univ. Brooklyn Campus, 617 W. 190th St. #2D, New York, NY 10040, gazcona326@gmail.com), Lupe Buten (Commun. Sci. and Disord., Long Island Univ. Brooklyn Campus, Yonkers, NY), and Lu-Feng Shi (Commun. Sci. and Disord., Long Island Univ. Brooklyn Campus, Brooklyn, NY)

This study explores how bilingual listeners' acceptable noise level (ANL) may be affected by the language of the signal, language of the masker, and talkers in the masker. ANL measures how much a listener tolerates background noise while listening to running speech. It differs from conventional speech recognition tasks in that it does not concern one's ability to comprehend speech. Hence, ANLs are expected to be relatively free of bilingual background. In this study, the signal (running passages from New York State Department of Motor Vehicles driving manual) was presented in two languages (English versus Spanish). The maskers (Auditec babbles) were manipulated in language (English versus Spanish) and number of talkers (four versus twelve). Additionally, three groups of 12 listeners participated in the study—monolingual English, Spanish-English bilingual, and Russian-English bilingual. A $3\times2\times2\times2$ mixed, repeated design was carried out with listener group as the between-subjects factor, and signal and masker language and number factors as the within-subjects factors. Preliminary findings reveal a non-significant effect for all four factors; however, a marginally significant four-way interaction (p = 0.048) invites group wise analysis. Results may help establish new clinical approaches in assessing listeners' speech perception difficulty in noise, regardless of language background.

5aPP14. An analysis of the acoustic characteristics of forest sounds. Doo-Heon Kyon and Myung-Jin Bae (Electronics Eng., Soongsil Univ., Hyeongnam Eng. Hall #1212,Sangdo-dong, Seoul 156-030, North Korea, kdhforce@gmail.com)

The purpose of this study is to define the general acoustic characteristics of forest sounds. Thus, large-scale measurements and analyses were

conducted throughout four seasons for three main mountainous areas in Korea. The results showed there were clear differences in the acoustic characteristics, depending on environments and seasons. As acoustic elements of a forest, there are sounds of water from waterfalls and streams and sounds from birds and insects, and even sounds caused by stepping on snow in winter or fallen leaves in autumn have effect on the seasonal acoustic characteristics. The frequency of forest sounds was about 20 dB(A) smaller than typical sounds occurring in a downtown area, and forest sounds showed flat frequency characteristics in general. Especially, it was investigated that the energy ratio of their ultra-high tone sound domains was only 0.1% of that of a downtown area, while the rate of forest sounds was about 50 times more than that of a downtown area, and there were large differences found in the other frequency bands. Based on the results above, such acoustic differences between forests and downtown areas might have effect on the human health in a long-term view.

5aPP15. Psychometric functions of sentence recognition in amplitude-modulated noises. Yi Shen, Nicole Manzano, and Virginia M. Richards (Dept. of Cognit. Sci., Univ. of California, Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697, shen.yi@uci.edu)

Although it is known that listeners can take advantage of the fluctuations in the temporal envelopes of background noises to improve the recognition of target messages, the amount of benefit is difficult to compare across listeners and experimental conditions. One solution to this difficulty is to estimate the psychometric functions for speech recognition instead of the speech reception threshold alone (i.e., the 50% correct point on the psychometric function). The current study utilized a rapid psychophysical procedure that enabled the robust estimation of the psychometric function for sentence recognition in noise using as few as 20 sentences. Using this procedure, sentence recognition was measured by presenting target sentences in amplitude-modulated noise maskers. In separate conditions, the target intensity (40 or 70 dB SPL) and the masker modulation rate (1–64 Hz) were systematically varied. Manipulating these two stimulus parameters influenced both the speech reception threshold and the slope of the psychometric function. Data collected from ten young, normal hearing listeners indicated that the fluctuating-masker benefit was much more evident at the higher target level and it exhibited non-monotonic dependencies on masker modulation rate.

5aPP16. Bayesian estimation of high-parameter models of the auditory filter. Yi Shen, Rajeswari Sivakumar, and Virginia M. Richards (Dept. of Cognit. Sci., Univ. of California, Irvine, 3151 Social Sci. Plaza, Irvine, CA 92697, shen.yi@uci.edu)

The Bayesian adaptive procedure proposed by Shen and Richards [J. Acoust. Soc. Am. **134**(2), 1134–1145 (2013)] was extended to allow the rapid estimation of auditory filters that were asymmetric about their peak frequencies. The estimation of the auditory-filter shape (five free parameters) was achieved using single Bayesian adaptive tracks of 150–200 trials (approximately 15 min to complete). During the experimental track, listeners detected a tonal signal presented with either simultaneous or forward maskers. Both types of maskers consisted of two bands of noises, one on each side of the signal frequency. The Bayesian adaptive procedure iteratively updated the parameter estimates following each experimental trial and determined the stimulus that would maximize the gain of information on the following trial. The stimuli were adaptively manipulated along three dimensions: the masker level and the spectral location of the upper and lower masker bands. The proposed procedure allowed the reliable estimation of the auditory-filter shape for naïve normal-hearing listeners. The model predictions replicated the known effect of the masker-signal simultaneity on the auditory filter.

5aPP17. Phase effects using chirp maskers. Niall A. Klyn, Yong Hee Oh, Evelyn Hoglund, 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)

Schroeder-phase maskers appear to interact with the phase curvature of the basilar membrane to produce significantly different amounts of masking depending on the direction of the instantaneous frequency sweep (Oxenham and Dau, 2001). Schroeder-phase maskers may not optimally compensate for the phase-curvature of the basilar membrane, which is not assumed to be linear over its length. This study presents a comparison of masking produced by harmonic complexes that differ only in the phase relationships of their components. Four "chirp" complexes were constructed according to Elberling *et al.* (2007), with both positive and negative signs, and were compared to Schroeder-phase signals (Smith *et al.*, 1986). Masking observed with pure-tones at three signal frequencies (1, 2, and 4 kHz) and with spectro-temporal burst signals (Hoglund *et al.*, 2013) is reported at two presentation levels (50 and 60 dB SPL). The results are discussed in the context of Schroeder-phase masking and current understanding of the phase-curvature of the human basilar membrane.

5aPP18. Reliable discrimination thresholds in 17 trials. Michael Stahl (Bioengineering, Northeastern Univ., 226 FR, Boston, MA 02115, stahl.m@meei.harvard.edu), Michael Epstein, and Mary Florentine (Speech-Lang. Pathol. and Audiol., Elec. and Comput. Eng., BioEng., Northeastern Univ., Boston, MA)

This study explores potential weaknesses of the Zippy Estimation by Sequential Testing (ZEST) psychophysical procedure as a means of acquiring rapid psychoacoustic measurements. Although ZEST is promising for this purpose, previous work used *a priori* knowledge to make optimal choices for initial assumptions, a best case scenario for ZEST's performance. Before the ZEST procedure can be employed as a clinical tool, an understanding of how it performs in the absence of a priori information is needed. Specifically, an investigator must choose: (1) the model psychometric function, (2) the starting level, and (3) the number of trials. The present study explores sensitivity to these choices when ZEST is employed in 2AFC, frequency-discrimination tasks. Data for six normal listeners were obtained for a wide range of initial conditions and compared with simulations. These data indicate that even when an inappropriate psychometric function is used, reliable thresholds can be obtained with only 17 trials when the starting level is within a factor of four times the listener's "true" threshold. These results suggest that ZEST combined with a 2AFC paradigm is a promising candidate for rapid and reliable assessment of listeners' discrimination thresholds. [Work supported by NIH/NIDCD 1R03DC009071.]

5aPP19. Effects of level roving and overall level on correlation change discrimination in naïve and trained listeners. Matthew J. Goupell and Mary E. Barrett (Hearing and Speech Sci., Univ. of Maryland, 0119E Lefrak Hall, College Park, MD 20742, goupell@umd.edu)

Sensitivity to interaural differences is remarkably variable across individuals, even highly trained listeners. In experiment 1, we measured sensitivity to changes in interaural correlation in 28 naïve listeners to determine the expected inter-individual variability for this task. Stimuli were 65-dB-A, 10-Hz narrowband noises with a 500- or 4000-Hz center frequency. Stimuli were tested either without level roving, or with ±5 dB of level roving in an effort to force listeners to attend to binaural width/movement cues rather than binaural loudness cues. At 500 Hz without level roving, the performance for the naïve listeners was much worse than the performance for the highly trained listeners previously reported in the literature. The addition of level roving significantly degraded performance, suggesting a close relationship between detection of decorrelation and loudness. At 4 kHz with and without level roving, none of the naïve listeners could perform the task. In experiment 2, we measured correlation change sensitivity as a function of overall level in trained listeners. Performance improved as overall level increased to about 65 dB-A, then worsened for more intense levels. The data will be discussed in terms of confusions between width and loudness cues, and in terms of Weber's law.

5aPP20. Gradual decay of auditory short-term memory. Samuel R. Mathias (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215, smathias@bu.edu), Christophe Micheyl (Starkey Hearing Res. Ctr., Berkeley, CA), and Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA)

Recent work suggests that once a representation of a visual object is committed to short-term memory, its precision remains fixed until it finally disappears without trace—a phenomenon called "sudden death." In the present study, we investigated whether auditory representations also experience sudden death. In three experiments, listeners discriminated either the pitches or loudnesses of pairs of tones separated in time by up to 10 s. The data were analyzed using a model that allowed us to estimate both the precision of pitch/loudness representations and the probability of sudden death as a function of the interstimulus interval from listeners' psychometric functions. Contrary to recent findings from vision, we found that auditory representations were no more likely to "die" after 10 s than 0.5 s; instead, they declined in precision (or "decayed") gradually over time. The results point to a qualitative difference between how auditory and visual representations are forgotten from short-term memory.

5aPP21. Do normal-reading listeners use "perceptual anchors" in frequency-discrimination tasks? Samuel R. Mathias (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215, smathias@bu.edu), Christophe Micheyl (Starkey Hearing Res. Ctr., Berkeley, CA), and Barbara Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA)

A controversial hypothesis about the origins of dyslexia states that dyslexic listeners are poor at frequency discrimination when the standard frequency is fixed from trial to trial because they find it difficult to use "perceptual anchors." This hypothesis tacitly assumes that normal listeners use perceptual anchors during fixed-frequency discrimination, which they cannot do when the standard frequency is random ("roved"). To test this assumption, we measured frequency-discrimination performance in normal listeners for tones separated by silent intervals (ISIs) of up to 10 s. Critically, in either the first or second half of the experiment (counterbalanced across listeners), the frequency of the first tone was the same on each trial ("fixed" conditions); in the other half, the frequency of this tone was random ("roved" conditions). The data were analyzed using a model that estimated the precision of frequency representations. In the roved conditions, frequency representations were less precise at longer ISIs, indicating that they experienced decay. By contrast, frequency representations experienced little decay in the fixed conditions. The results are consistent with the assumption that normal listeners used perceptual anchors in the fixed conditions because, by definition, perceptual anchors should be stored in long-term memory and therefore less susceptible to decay.

5aPP22. Induced loudness reduction as a function of inducer level. Michael Epstein and Mary Florentine (Speech-Lang. Pathol. and Audiol., Elec. and Comput. Eng., Bioengineering, Northeastern Univ., 360 Huntington Ave., 106A FR, Boston, MA 02115, m.epstein@neu.edu)

Induced loudness reduction (ILR) is a phenomenon that occurs when the loudness of a stimulus (test tone) is reduced by the presence of a proceeding sound (inducer tone) that is presented at a level higher than the stimulus. This effect depends on a number of parameters, including inducer level. Whereas a number of studies have examined the effects of inducer level, there is no comprehensive data set examining the effects of a wide range of inducer levels on a wide range of test-tone levels. The present study fills this void by examining the effects of ILR for 500-Hz inducers when matching affected 500-Hz test tones at fixed levels of 5 dB SL and 20, 40, 60, 80, and 90 dB SPL with unaffected 2500-Hz level-adjusted comparison tones. Inducers primarily had an effect on test tones lower than the inducer level. Moderate-level tones exhibited the most ILR. This is consistent with prior studies examining single inducer levels. In addition, low- and moderate-level inducers appear to slightly increase the loudnesses of sounds at higher levels, indicating that the loudnesses of sounds at virtually every sound level within the dynamic range of hearing can be altered by ILR. [Work supported by NIH/NIDCD 1R03DC009071.]

5aPP23. Factors affecting auditory streaming of random tone sequences. An-Chieh Chang, Inseok Heo, Jungmee Lee, Christophe Stoelinga, and Robert Lutfi (Commun. Sci. and Disord., Univ. of Wisconsin - Madison, 1975 Willow Dr., Madison, WI 53706, achang5@wisc.edu)

As the frequency separation of A and B tones in an ABA_{ABA} tone sequence increases the tones are heard to split into separate auditory streams (fission threshold). The phenomenon is identified with our ability to 'hear out' individual sound sources in natural, multisource acoustic environments. One important difference, however, between natural sounds and the tone sequences used in most streaming studies is that natural sounds often vary unpredictably from one moment to the next. In the present study, fission thresholds were measured for ABA_{ABA} tone sequences made more or less predictable by sampling the frequencies, levels or durations of the tones at random from normal distributions having different values of sigma (0–800 cents, 0–8 dB, and 0–40 ms, respectively, for frequency, level, and duration). Frequency variation on average had the greatest effect on threshold, but the function relating threshold to sigma was non-monotonic; first increasing then decreasing for the largest value of sigma. Differences in the sigmas for A and B tones tended to reduce thresholds, but covariance in the A and B tones had little effect. The results suggest that the principles of perceptual organization underlying streaming may differ for predictable and unpredictable tone sequences.

5aPP24. Bidirectional audiovisual interactions: Evidence from a computerized fishing game. Seth Bensussen, Kenny F. Chou (Dept. of Biomedical Eng., Boston Univ., 677 Beacon St., Boston, MA 02215, sethbens@bu.edu), Lenny A. Varghese (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA), Yile Sun (Volen National Ctr. for Complex Systems, Brandeis Univ., Waltham, MA), David C. Somers (Dept. of Psych., Boston Univ., Boston, MA), Robert Sekuler (Volen National Ctr. for Complex Systems, Brandeis Univ., Waltham, MA), and Barbara G. Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., Boston, MA)

We used a specially designed computer game to examine behavioral consequences of audiovisual integration. Target stimuli (animated fish swimming across the computer screen) were modulated in size and/or emitted an amplitude-modulated sound. Modulations, visual or auditory, were at 6 or 7 Hz (corresponding to "slow" and "fast"). In one game, subjects were instructed to categorize successive fish as "slow" or "fast" based on the auditory modulations; in another game, they categorized fish based on visual modulation rate. In each game, subjects were instructed to ignore input from the task-irrelevant modality. In each game, modulations could be (1) present only in the modality of interest, (2) present and matching in both modalities, or (3) present but mismatched between modalities. While reaction times were similar across games, accuracy was highest when auditory modulation was the basis for categorizing fish. Accuracy and reaction times improved when cross-modal modulation rates matched, and worsened when modulation rates conflicted. Additionally, accuracy was more strongly affected by between-modality congruence/incongruence when subjects attended to visual modulations than when they attended to auditory ones. Results indicate that audiovisual integration is not entirely under volitional control, and that competition between sensory modalities adversely impacts perception in dynamic environments.

5aPP25. Contribution of detailed parts around talker's mouth for audio-visual speech perception. Shuichi Sakamoto (Res. Inst. of Elec. Commun., Tohoku Univ., 2-1-1 Katahira, Aoba-ku, Sendai, Miyagi 980-8577, Japan, saka@ais.riec.tohoku.ac.jp), Gen Hasegawa (Graduate School of Information Sci., Tohoku Univ., Sendai, Japan), Toru Abe (CyberSci. Ctr., Tohoku Univ., Sendai, Japan), Tomoko Ohtani, Yôiti Suzuki (Res. Inst. of Elec. Commun., Tohoku Univ., Sendai, Japan), and Tetsuaki Kawase (Graduate School of Biomedical Eng., Tohoku Univ., Sendai, Japan)

In this study, the relationship between speech intelligibility and effects of the parts around talker's mouth was investigated based on the results of audio-visual speech intelligibility tests. As the stimuli, nonsense tri-syllables speech sounds were combined with three kinds of moving image of the talker's face: original face, circumference of the lips (mouth part extracted from

the original face), and masked lips (face without the mouth). The extracted area around the mouth was varied as a parameter in the circumference-of-the-lips conditions. Generated stimuli were presented with speech spectrum noise to the participants. The results showed that intelligibility scores of several phonemes (/h/, /m/, /w/, /b/, /p/) were increased by adding the visual information. Moreover, there was no significant difference between the score of the original-face condition and that of the circumference-of-the-lips condition. Interestingly, significant difference was observed between the score of the audio-only condition and that of the masked-lips-condition. These results imply that not only a mouth but also areas around the mouth without mouth itself provide rich information for speech understanding. Effects of the mouth edges on speech intelligibility are being examined as a next step.

5aPP26. Auditory attention in a dynamic scene: Behavioral and electrophysiological correlates. Hannah R. Goldberg, Inyong Choi, Lenny A. Varghese, Hari Bharadwaj, and Barbara G. Shinn-Cunningham (Auditory Neurosci. Lab, Boston Univ., 677 Beacon St., Boston, MA 02215, hannahrae.goldberg@gmail.com)

The ability to direct and redirect selective auditory attention varies substantially across individuals with normal hearing thresholds, even when sounds are clearly audible. We hypothesized that these differences can come from both differences in the spectrotemporal fidelity of subcortical sound representations and in the efficacy of cortical attentional networks that modulate neural representations of the auditory scene. Here, subjects were presented with an initial stream from straight ahead and a second stream (from either left or right), each comprised of four monotonized consonant-vowel syllables. Listeners were instructed to report the contents of either the first stream (holding attentional focus) or the second stream (switching attentional focus). Critically, the direction of the second stream informed subjects whether to hold or to switch attention. Pilot results suggest that when the lateral angle of the second stream is small, task performance is linked to subcortical encoding fidelity of suprathreshold sound (as measured using brainstem frequency-following responses obtained separately in the same subjects). Using a paradigm that allows simultaneous collection of behavioral measures, FFRs, and cortical responses, here we test whether differences in top-down attentional control explain subject variability when the second stream's lateral angle is large and coding fidelity does not limit performance.

5aPP27. Behavioral and neural measures of auditory selective attention in blast-exposed veterans with traumatic brain injury. Scott Bressler, Inyong Choi, Hari Bharadwaj, Hannah Goldberg, and Barbara Shinn-Cunningham (CompNet, Boston Univ., 677 Beacon St., Rm. 304, Boston, MA 02215, bressler@bu.edu)

It is estimated that 15–20% of veterans returning from combat operations have experienced some sort of traumatic brain injury (TBI), a majority of which are the result of exposure to blast forces from military ordinance. Many of these veterans complain of complications when understanding speech in noisy environments, even when they have normal or near normal audiograms. Here, ten veterans diagnosed with mild TBI performed a selective auditory attention task in which they identified the shape of one of three simultaneously occurring ITD-spatialized melodies. TBI subject performance was significantly worse than that of 17 adult controls. Importantly, the veterans had hearing thresholds within 20 dB HL up to 8 kHz and brainstem responses indistinguishable from those of the controls. Cortical response potentials (measured on the scalp using electroencephalography) from correctly identified trials showed weaker attention-related modulation than in controls. These preliminary results suggest blast exposure damages cortical structures responsible for controlling selective auditory attention (modulating sensory representations in cortex), and represent a step toward developing novel diagnostic methods to assess functional consequences of specific patterns of blast TBI damage in individual patients.

5aPP28. A bilingual advantage in the irrelevant speech effect. Josh Dorsi (Psych., UC Riverside, 34875, Winchester, NY 92596, jdors002@ucr.edu), Dominique Dominique Simmons, Theresa Cook, Lawrence Rosenblum (Psych., UC Riverside, Riverside, CA), and Oksana Laleko (English, SUNY New Paltz, New Paltz, NY)

The bilingual advantage refers to superior performance by multilingual over monolingual individuals on a variety of linguistic, and non-linguistic cognitive tasks. Tasks that include the bilingual advantage are ones that involve attention control, such as the Simon or Stroop tasks, but not tasks that involve short-term memory (Bialyskok, 2009). The Irrelevant Speech Effect (ISE) is the finding that serial recall accuracy of visual list items declines when task irrelevant speech backgrounds are present (relative to the disruption of white noise backgrounds: Colle and Welsh, 1976; Jones and Macken, 1993). The theories that have been offered to account for ISE are of two kinds, those that assume recall disruption is a function of limited memory, and those that assume it is the result of diffused attention (Elliot, 2002). As the bilingual advantage is found in tasks of attention, but not those concerning memory, a bilingual advantage in the ISE would support an attention based theory. In order to test this prediction, bilingual and monolingual subjects performed serial recall tests against speech and white noise backgrounds. Preliminary results suggest that the size of ISE is substantially smaller in bilingual, than it is in monolingual participants supporting attention-based theories of the ISE.

5aPP29. Air conduction and bone conduction in aging mice. David Chhan (Speech and Hearing BioSci. and Technol., Harvard-MIT Health Sci. and Technol., 1200 Massachusetts Ave., Apt 53E, Cambridge, MA 02138, dchhan@mit.edu), Melissa L. McKinnon (Eaton Peabody Lab., Massachusetts Eye and Ear Infirmary, Boston, MA), and John J. Rosowski (Dept. of Otology and Laryngology, Harvard Med. School, Boston, MA)

Umbo velocity and Auditory Brainstem Response (ABR) measurements in aged mice suggest that there are functional changes in both the inner ear and middle ear for frequencies from 5 to 13 kHz (Doan *et al.*, 1996). In this work, we use a combination of air conduction (AC) and bone conduction (BC) stimulation to better quantify the middle ear contribution to age-related hearing loss seen in mice. ABRs were recorded with AC and BC stimuli from BALB/c mice of four different age groups (1, 2, 8, and 12 months); mice of this strain are widely used as models for age-related hearing loss. Results show the threshold stimulus levels for both AC and BC increase as the mice get older, consistent with age-related hearing loss. At frequencies below 12 kHz, the age-related changes in thresholds for all age groups are similar for both stimuli: the AC-BC difference (the air-bone gap) is not statistically significant. This suggests in this frequency range, the hearing loss is primarily sensorineural. At 16 kHz, the air-bone gaps of the two oldest groups are statistically significant suggesting the middle ear contributes to the hearing loss. Thresholds at higher frequencies were not measurable in the two oldest groups.

5aPP30. Can frequency discrimination be indexed by electrophysiological measures. Wen-Jie Wang, Brett A. Martin, Glenis R. Long (Ph.D. Program in Speech-Language-Hearing Sci., The Graduate Ctr., City Univ. of New York, 365 Fifth Ave., New York, NY 10016, wwang2@gc.cuny.edu), and Chin-Tuan Tan (Dept. of Otolaryngol., New York Univ. School of Medicine, New York, NY)

Acoustic change complex (ACC) responses to quadratic frequency glides following a steady pure tone were obtained from eight normal-hearing listeners. The base frequencies were 500 and 1000 Hz. The glides either increased or decreased in frequency by 200 Hz with durations of 50, 100, and 200 ms. The N1 response to stimulus onset and glide onset were evaluated to determine if the elicited ACC was associated with the frequency change in the glides. In contrast to the onset N1, the ACC N1 was delayed.

This delay was used to index the time for frequency glides to reach a potentially detectable frequency change. ACC N1 latency was longer for glides of longer duration. However, the instantaneous frequency at ACC N1 peak latency was relatively constant regardless of glide duration. The difference in ACC N1 latency across glide durations was diminished when expressed as stimulus frequency, ratio of frequency change to base frequency, or associated place shift along the basilar membrane. There was also an effect of base frequency. ACC N1 latency was evoked when the percentage change from the base frequency exceeded ~0.2%. There was no clear evidence that the ACC is dependent solely on the duration or rate of frequency change.

5aPP31. A computational study on different flow patterns around the inner hair cell stereocilia. Srdjan Prodanovic, Sheryl M. Gracewski (Dept. of Mech. Eng., Univ. of Rochester, 383 Quinby Rd., Rochester, NY 14623, s.prodanovic@rochester.edu), and Jong-Hoon Nam (Dept. of Mech. Eng., Dept. of Biomedical Eng., Univ. of Rochester, Rochester, NY)

The mechano-transduction of the mammalian cochlea occurs in the micro-fluid domain between the tectorial membrane and the reticular lamina called the subtectorial space. The subtectorial fluid bathes the bundled stereocilia of the inner hair cells (IHCs). These cells are responsible for the onset of neural impulses in the auditory nerve fibers. Despite the generally accepted postulation that the IHC stereocilia are deflected by shear flow between the two layers, there have been suggestions that other flow modes exist besides the shear flow. We developed a computational model of fluid dynamics in the subtectorial space. The model simulates IHC mechano-transduction excited by different flow patterns. In order to compare different modes of fluid dynamical stimulation, the power efficiency of IHC mechano-transduction was introduced (dissipated power normalized by IHC mechano-transduction current). Besides different flow patterns, the effect of mechanical parameters (such as the gap size between the stereociliary tip and the tectorial membrane, stereociliar bundle stiffness) were investigated. The results demonstrate that the power efficiency for the IHC mechano-transduction depends on the flow pattern in the subtectorial space.

5aPP32. Effect of spatial discontinuity on short-term memory for melodic sequences. Audrey S. Wang, Lenny A. Varghese, Samuel R. Mathias, and Barbara G. Shinn-Cunningham (Ctr. for Computational Neurosci. and Neural Technol., Boston Univ., 677 Beacon St., Boston, MA 02215, Audrey_Wang@buacademy.org)

We investigated whether spatially separating segments of a tone sequence split the sequence into separate perceptual streams, and whether it affected how pitch information was stored in memory. Listeners heard a sequence of six 300-ms complex tones presented either to a single ear, or with the first three and last three tones presented to different ears. They reported whether a subsequent monaural probe tone matched a tone in the original sequence, based solely on pitch (disregarding the ear to which the probe was played). Overall, performance was higher when the probe tone matched either the first or one of the last two tones in a target compared to when the probe matched more interior target elements. For single-ear target sequences, presentation of the probe in the opposite ear diminished performance compared to when it was presented in the same ear as the target. Spatially splitting the target sequence had very little effect. The results suggest that the tone sequences were perceived as a single stream, irrespective of whether there was spatial discontinuity. However, a probe in an unexpected location confused listeners, perhaps causing a spatial attention shift that diminished the already-weak memory representation of the interior tones' pitches.

5aPP33. Auditory cortex membrane potential dynamics in mice during challenging and ethologically relevant natural soundscapes. Mathew J. McGinley, Gregg A. Castellucci, and David A. McCormick (Neurobiology, Yale Univ., 333 Cedar St., New Haven, CT 06510, matthew.mcginley@yale.edu)

Natural sound environments present numerous spectro-temporally complex and highly non-linear sound sources that an animal must assess for relevance. The auditory cortex is thought to be important for the execution of challenging sensory processing tasks. Yet, recordings in the auditory cortex have largely been studied in response to simple or synthetic sounds, or single well-isolated natural sound patterns such as con-specific vocalizations. In particular, membrane potential s in auditory cortex have only been recorded in response to simple stimuli, to which they respond with large rapid depolarization from a stable hyperpolarized baseline voltage. To help determine the dynamics of membrane potential fluctuations in natural conditions, we played natural soundscapes to head-fixed awake mice on a cylindrical treadmill while recording membrane potentials of pyramidal neurons the primary auditory cortex of mice. Ambient sounds were recorded from forests or meadows, particularly near creeks and during sunrise or sunset when numerous species were active. Sound frequencies encompassing the entire hearing range of mice (1–250 kHz) were recorded using a AviSoft UltraSoundGate recording system with CM16/CMPA microphone.

5aPP34. Parameter fitting of a lumped parameter middle ear model. Charlsie Lemons and Julien Meaud (Mech. Eng., Georgia Inst. of Technol., 1035 Hampton St, Atlanta, GA 30318, charlsielem@gmail.com)

Measurements of otoacoustic emissions in the ear canal are affected by the forward and reverse pressure transfer functions and the reverse middle ear impedance. Thus, a comprehensive middle ear model should reflect each of these measurements of middle ear function using a single set of parameters. A middle ear circuit model consisting of a lumped parameter representation of the ossicular chain coupled to a transmission line model of the eardrum was selected (O'Connor and Puria, JASA, 2008). In the original model, the parameters were fit solely to measurements of the magnitude and phase of the stapes velocity relative to the ear canal pressure in the forward direction. In this paper, the model parameters are optimized through a non-linear least square regression method so that the model's forward pressure transfer function, reverse pressure transfer function, and reverse middle ear impedance agree with experimental data for the human middle ear from 0.1 to 10 kHz. This parameter fitting procedure is repeated using experimental data for the guinea pig and gerbil middle ear. Differences in model parameters between the three species are discussed.

5aPP35. Aligning digital holography images of tympanic membrane motion. Jeremie Guignard, Jeffrey T. Cheng, Michael E. Ravicz, and John J. Rosowski (Eaton-Peabody Lab., Massachusetts Eye and Ear Infirmary, 243 Charles St., Boston, MA 02114-3096, jeremie_guignard@meei.harvard.edu)

Modern studies of sound-induced tympanic membrane (TM) motion include digital holography and optical-coherence tomography. These techniques generate high-definition maps of the motion of human and animal TMs that define and quantify spatial patterns of motion. In order to compare such patterns quantitatively between multiple specimens, a common coordinate system is necessary. Differences in the relative position of the sensor and the measured sample induce differences in magnification, shape, and location of the resulting TM image. A registration algorithm was implemented as follows: (1) The plot of the phase of motion at low frequencies was filtered with a moving 2D variance filter to differentiate the membrane from the bone in both reference and moving images (a process called segmentation). (2) A routine calculated the covariance matrix of the segmented images to measure its second central moment. (3) The images were aligned and warped on the basis of their centroid and moment. The method is about ten times faster than state-of-the-art intensity based registration, and the average squared errors of the pixel intensities after registration are not statistically different. The method is an efficient strategy to align TM surface images.

5aPP36. Investigating perceptual structure of concert hall ambiances. Aparna Kanakanahalli Nagendra and SungYoung Kim (TeleCommun. Eng. Technol., Rochester Inst. of Technol., 148 Crittenden Way Apt., #6, Rochester, NY 14623, axk9226@rit.edu)

With the advent of ultra high definition (UHD) video, a new sound reproduction system incorporating height channel to enhance audio-visual interaction is getting researchers' attention. To better understand influence of the contents for the height channel(s) on perceived sound quality, the authors have conducted a listening test wherein listeners compared various concert hall ambiances. The authors first recorded the ambiances of a hall at

175 positions (in various heights and positions) using a MIDI-controlled piano. Subsequently we investigated interrelation of ambiances with subordinate objectives: (1) reveal perceptual space of the ambiances and (2) unfold latent yet salient factors. We selected 10 ambiances placed at three-dimensional equidistant from the source as the stimuli. Fifteen musicians participated in a subjective evaluation where they reported perceived dissimilarity between two randomly presented stimuli. The dissimilarity matrices were then analyzed via INDSCAL and the results showed that the stimuli could be coordinated along two bases which corresponded to distance from the source and spectral kurtosis of the stimuli respectively. Finding those salient bases will help assist spatial audio researchers in rendering height ambiances with smaller yet relevant subset of ambiances.

5aPP37. Effects of spectral degradation on attentional modulation of auditory responses to continuous speech. Ying-Yee Kong (Dept. of Speech Lang. Pathol. & Audiol., Northeastern Univ., 226 Forsyth Bldg., 360 Huntington Ave., Boston, MA 02115, yykong@neu.edu), Ala Mullangi (BioEng. Program, Northeastern Univ., Boston, MA), and Nai Ding (Dept. of Psych., New York Univ., New York, Massachusetts)

Auditory attention enhances phase-locked responses to the temporal envelope of the attended speech stream in a competing background. This study investigates how cortical responses are affected by spectral degradation and the extent to which low-frequency fine-structure cues facilitate sound segregation. Two competing speech streams were presented diotically to normal-hearing subjects. Each subject was tested with unprocessed speech, vocoder speech (8 to 64 channels), and six-channel high-frequency vocoder + low-pass filtered speech simulating electric-acoustic stimulation (EAS). Ongoing EEG responses were measured in each condition. Cross-correlation between speech envelope and EEG responses was calculated at different time lags. For unprocessed speech, the cross-correlation function showed opposite signs of correlations between the attended and unattended speech, supporting the neural mechanism of suppression of the competing speech stream. This suppression mechanism, however, was only evident for the 32- and 64-channel vocoder conditions, but not for the 8- and 16-channel conditions. This indicates that greater frequency resolution is required for sound segregation. For EAS, the pattern of cortical responses was similar to that of the eight-channel vocoder condition, although speech intelligibility was higher in EAS than vocoder speech. Consistent with previous psychoacoustic evidence, these neural results further argue against the segregation mechanism for EAS benefit.

FRIDAY MORNING, 9 MAY 2014 — 553 A/B, 9:00 A.M. TO 12:00 NOON

Session 5aSA

Structural Acoustics and Vibration: Recent Advances in Structural Acoustics and Vibrations

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

Michael A. Jandron, Cochair
Naval Undersea Warfare Ctr., Code 8232, Bldg. 1302, Newport, RI 02841

Contributed Papers

9:00

5aSA1. Efficient analysis of dynamic coupling between modifications to complex systems. Andrew Wixom and J. Gregory McDaniel (Mech. Eng., Boston Univ., 110 Cummington Mall, Boston, MA 02215, awixom@bu.edu)

This paper outlines a strategy for identifying and exploiting the lack of dynamic coupling between potential modifications to a vibroacoustic system. Often a designer develops a set of modifications and wishes to determine those that perform the best relative to some metric, in particular, we consider performance metrics that rely on the steady state forced response of the structure over a frequency band of interest. In this case, the effects of coupling are shown to appear as a force residual and therefore may be neglected if sufficiently small. When modifications are uncoupled from one another, the computational cost of analyzing all possible modified systems can be greatly reduced. For example, if up to 4 of 7 potential modifications are allowed to be applied to a system, there are 98 possible modified systems that need to be analyzed to determine the optimal combination. However if all of the modifications are found to be uncoupled, then only seven modified systems need to be analyzed to generate the results for all possible modified systems. An example problem demonstrates the benefits of uncoupling between modifications.

9:15

5aSA2. Resonant vibration control of a cylindrical shell using distributed tuned mass dampers. Christopher Page (Noise Control Eng., Inc., 799 Middlesex Turnpike, Billerica, MA 01821, c.page@noise-control.com), Peter Avitabile, and Christopher Niezrecki (Structural Dynam.. & Acoust. System Lab., Univ. of Massachusetts - Lowell, Lowell, MA)

Often structure borne noise has a deleterious effect on the performance and aesthetics of many commercial and military systems. Solutions to structural noise problems can be broadly classified into three categories: active, semi-active, and passive. With ever increasing improvements in electronics and computing, much of the recent research emphasizes active and semi-active solutions. However, passive solutions remain favorable in practice as they are often less costly to implement and less imposing on the system architecture. The tuned mass damper (TMD) is a common passive approach. Tuned mass dampers are generally used to affect a single undesirable resonance or forcing frequency. In order to affect multiple resonances multiple tuned absorbers are typically required, causing the design and implementation to be prohibitively difficult. This paper develops the concept of a modally enhanced dynamic absorber (MEDA) as a dynamic design philosophy, whereby the mass of peripheral equipment can be utilized and employed in attenuating the structural response of a larger system in some

desirable way. The concept is demonstrated numerically and experimentally on a steel cylindrical shell. The importance of TMD attachment location with respect to the mode being de-tuned, robustness of TMD damping, and the effect of TMD mistuning is studied.

9:30

5aSA3. Transient response of flat-panel distributed mode loudspeakers.
David Anderson, Stephen Roessner, and Mark F. Bocko (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, dander22@hse.rochester.edu)

The acoustic impulse response of a loudspeaker must have short rise and decay times to provide accurate reproduction of signals with fast transients, such as human speech. In a flat panel distributed mode loudspeaker (DML), the transverse bending waves that are the primarily source of acoustic radiation, propagate across the panel at finite velocity. Therefore, regions of the panel more distant from the panel driving point(s) radiate sound at later times, which broadens the DML's acoustic impulse response. Furthermore, bending wave propagation in a rigid panel is dispersive, with a wave velocity that is proportional to the square root of the frequency, which creates additional distortions of signals with fast transients. Mechanical simulations and scanning laser vibrometer measurements of glass panel DML's clearly illustrate the delayed radiation and dispersion distortion mechanisms. Furthermore, measurements of the acoustic impulse response of DML's are in agreement with simulations, which provides additional evidence that the identified mechanisms are a major source of audio distortion in DML's.

9:45

5aSA4. Audio frequency response of glass flat panel distributed mode loudspeakers.
Stephen Roessner, David Anderson, and Mark F. Bocko (Dept. of Elec. and Comput. Eng., Univ. of Rochester, Rochester, NY 14627, stephen.roessner@rochester.edu)

The development of glass flat panel distributed mode loudspeakers (DML's) will enable compelling new applications, such as flat panel displays that double as loudspeakers, or in architectural and automotive applications, window glass that can produce sound or actively cancel environmental noise. In this paper we discuss the frequency response of glass flat-panel distributed mode loudspeakers. The density of panel bending modes, their quality factors (Q's), and the modal radiation efficiencies determine a DML's frequency response. For example, a 0.55 mm thick cover glass for a 55 in. television has more than 10 000 bending modes in the audible frequency range. The density of modes increases with frequency and above a threshold frequency, determined by the panel dimensions and the mode Q's, the mode spacing becomes less than the width of the individual modes, so the overall frequency response approaches the smooth, flat response of an ideal pistonic loudspeaker. However, below this frequency, discrete modes produce prominent peaks in the frequency response. We present simulated and measured glass panel frequency responses, including the effect of a thin layer of trapped air behind a DML panel, and we discuss the need for high internal friction (low Q) glass.

10:00

5aSA5. An analysis of the weighted sum of spatial gradients (WSSG) control metric in active structural acoustic control.
Yin Cao, Scott D. Sommerfeldt (Dept. of Phys. and Astronomy, Brigham Young Univ., N225 ESC, Provo, UT 84602, caoyfive@gmail.com), Jonathan D. Blotter (Dept. of Mech. Eng., Brigham Young Univ., Provo, UT), and Pegah Aslani (Dept. of Phys. and Astronomy, Brigham Young Univ., Provo, UT)

The choice of the control metric used and how it is measured is an important factor in active structural acoustic control (ASAC) and has been a topic of research for many years. In ASAC, the objective is to minimize the radiated power, but this metric is difficult to implement in practice. Previous research has identified a control metric named the weighted sum of spatial gradients (WSSG) that has demonstrated effectiveness in attenuating the radiated power from a plate. This metric only requires local vibration information on the plate and hence is easier to implement than previous control metrics. It is also largely insensitive to boundary conditions. In addition, ASAC using WSSG has been demonstrated to be generally more effective than using volume velocity as the control metric. This paper demonstrates that WSSG closely approximates the ideal of using the radiated sound power as the control objective. It will be shown that the weights used to implement WSSG are also very important, as they directly impact the control effectiveness. Both theoretical and experimental results show that optimum weights exist to implement WSSG. These results suggest that WSSG has the potential ability to be used in more practical and complex structures.

10:15–10:30 Break

10:30

5aSA6. Dynamic response of cross-linked random fiber networks.
Sahab Babaee (SEAS, Harvard Univ., 29 Oxford St., Pierce Hall 410, Cambridge, MA 02138, sbabaee@fas.harvard.edu), Ali Shahsavari, Catalin Picu (Dept. of Mech. Eng., Rensselaer Polytechnic Inst., Troy, NY), and Katia Bertoldi (SEAS, Harvard Univ., Cambridge, MA)

Systems composed of fibers are ubiquitous in the living and artificial systems, including cartilage, tendon, ligaments, paper, protective clothing, and packaging materials. Given the importance of fiber systems, their static behavior has been extensively studied and it has been shown that network deformation is nonaffine for compliant, low-density networks and affine for stiff, high-density networks. However, little is known about the dynamic response of fibrous systems. In this work, we investigate numerically the propagation of small-amplitude elastic waves in these materials and characterize their dynamic response as a function of network parameters.

10:45

5aSA7. Optimization of spherical numerical integration methods in reconstruction of sound field with spherical microphone array.
Minzong Li, Jiangming Jin, and Huancai Lu (College of Mech. Eng., Zhejiang Univ. of Techonolgy, 18#, Chaowang Rd., Hangzhou 310014, China, liminzong_svlab@163.com)

Choosing an effective spherical numerical integration method is one of the key problems in reconstruction of incident sound field by using spherical microphone array based on spherical near-field acoustic holography (SNAH). Many different spherical numerical integration methods are reported, such as I.H.Sloan, Spherical-t design, J.Fliege, etc., each method applies different distributions of discrete positions and integral weights in the design of spherical microphone array. In the paper, the accuracies of sound pressure reconstruction results based on SNAH with different spherical numerical integration methods are examined and compared. It is found that spherical-t integration method can provide more options of discrete points, simple algorithmic due to equal weight of each point, and the same or higher precision of sound pressure reconstruction. Meanwhile, the expansion terms of reconstructed sound pressure has to be truncated at an optimal number in order to minimize the impact of spherical numerical integration error, the approach to choosing the optimal truncated term number with respect of the number of microphones in spherical numerical integration is discussed in details.

11:00

5aSA8. Patch nearfield acoustic holography based visualization of spatial distribution of sound quality objective parameters.
Jiangming Jin, Fang Yuan, and Huancai Lu (College of Mech. Eng., Zhejiang Univ. of Techonolgy, 18#, Chaowang Rd., Hangzhou 310014, China, jjm@zjut.edu.cn)

Psychoacoustic research reveal that sound pressure level reduction in cabin does not mean improvement of human's subjective auditory perception. This paper presents Patch Nearfield Acoustics Holography-Sound Quality (Patch NAH-SQ) combined analysis method with the advantages of NAH and SQ. This methodology breaks the limit of measurement aperture of conventional NAH and can visualize of spatial distribution of sound pressure and sound quality objective parameters of entire sound field by using a small number of microphones. From the 3D images of spatial distribution of sound pressure and sound quality objective parameters, the sound sources which most contribute to human's auditory perception can be located. In this paper, the impact of the choice of parameters such as overlap ratio of

Patch holographic surface, SNR, stand-off distance and frequency on reconstruction error of sound pressure and sound quality objective parameters is examined. Finally, the experiment of Patch NAH-SQ methodology proposed is validated in anechoic camber, the results are presented and analyzed.

11:15

5aSA9. A fully coupled finite element-boundary element vibro-acoustic analysis for laminated composite structures with enclosed acoustic cavities. Atanu Sahu, Arup Guha Niyogi (Dept. of Civil Eng., Jadavpur Univ., 188 Raja S.C. Mallik Rd., Kolkata, West Bengal 700032, India, sahuatanu@daad-alumni.de), Michael Rose (Inst. of Composite Structures & Adaptive Systems (Adaptronic Devision), German Aerosp. Ctr. (DLR), Braunschweig, Germany), and Partha Bhattacharya (Dept. of Civil Eng., Jadavpur Univ., Kolkata, India)

A major contribution of aircraft cabin noise is due to the vibrating airframe structure in which the use of lightweight composite materials is quite common. Due to the strong influence of such vibrating structures on the enclosing fluid, such as air, and vice-versa, a fully coupled analysis involving both the systems is necessary in many cases. Hence the present work details a fully coupled vibro-acoustic analysis of such type of flexible cavity subjected to the external mechanical excitation based on a coupled finite element (FE) and boundary element (BE) approach. The structural mode shapes and eigen frequencies are computed using the FE free vibration analysis of *in-vacuo* structures, which are subsequently coupled through the mobility equation with the acoustic part modeled with the BE method. The developed approach is also well compared with the fully coupled FE approach but the advantage derived here is that it is sufficient to model each domain independently thus reducing the problem dimension. The discussed method ensures full coupling between the two systems considering the cavity acoustic back pressure and also enables a fully coupled analysis of a double-walled structure attached to an acoustic cavity which resembles a true aircraft cabin.

11:30

5aSA10. Optimal research on volume velocity-matching wave superposition method based on tripole. Shaowei Wu and Yang Xiang (School of Energy and Power Eng., Wuhan Univ. of Technol., Peace Ave., Wuhan, Hubei Province, No. 1040, Wuhan, Hubei 430063, China, thinkwsw@qq.com)

A volume velocity-matching wave superposition method for sound prediction of vibrating structure based on tripole is proposed. In the method, the position of the equivalent sources has similar geometry shape with radiating structure using a scale coefficient K. The calculating cut-off frequency of structure under a meshing pattern is predicted by setting 1.5% volume velocity relative error limit using surface vibration velocity. The cut-off frequency is defined as the maximum value of all wavenumber ka, at which the relative error just exceeds 1.5% when K iterates through (0,1) with step K. Then, for calculating frequency range [k1,k2], the optimal position of the equivalent sources, in which the relative error of volume velocity is minimum, is determined in the calculating cut-off frequency range by searching K in set $\{K|K, ka \geq k2\}$. The transfer matrix between pressure and surface volume velocity is constructed in the optimal position by using tripole as equivalent source. After that, the sound radiation of structure is predicted to get high precision prediction. The results of numerical simulation show that the method is good at predicting the sound radiation of structure and the predicting error is very low within the cut-off frequency range once the optimal position is determined.

11:45

5aSA11. Guided modes with multiple zero-group-velocity points in fluid-filled cylindrical pipes. Hanyin Cui, Weijun Lin, Hailan Zhang, Xiuming Wang (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., No. 21, Beisihuanxi Rd., Beijing 100190, China, cuihanyin@mail.ioa.ac.cn), and Jon Trevelyan (School of Eng. and Computing Sci., Durham Univ., Durham, United Kingdom)

It is known that guided modes in isotropic hollow cylinders exhibit backward wave propagation with negative group velocity. And interferences between backward and forward waves generate zero-group-velocity (ZGV) resonances with a finite wavenumber but vanishing group velocity. These ZGV resonances can be applied for non-destructive evaluation (NDE) of hollow pipes. In this paper, influences of a fluid-loading on ZGV resonances in pipes are studied for the possible application of integrity inspections of oil transportation pipelines. From numerically simulated frequency-wavenumber spectra of axisymmetric guided modes, in addition to the backward mode with a single ZGV point, certain branches change the sign of their slopes for twice (i.e., two ZGV points in one branch). Such multiple ZGV modes might be caused by the strong repulsion between the backward mode with a single ZGV point that is propagating in the hollow pipe and a number of longitudinal modes in the fluid cylinder. It is found that, from wave structure analyses, ZGV points correspond to relatively large displacement amplitudes at the pipe's inner and outer interfaces. It indicates that guided modes with multiple ZGV points can be sensitive to the surface features of fluid-filled pipes, which is useful for NDE application.

Session 5aSC

Speech Communication: Speech Perception II (Poster Session)

Neal P. Fox, Chair
CLPS Dept., Brown Univ., Box 1821, Providence, RI 02912

All posters will be on display from 8:00 a.m. to 12:00 noon. To allow contributors the opportunity to see other posters, author of odd numbered papers will be at their posters from 8:00 a.m. to 10:00 a.m. and authors of even numbered papers will be at their posters from 10:00 a.m. to 12:00 noon.

Contributed Papers

5aSC1. Infants' perception of source size in vowel sounds. Matthew Masapollo, Linda Polka (McGill Univ., 1266 Pine Ave. West, Montreal, QC H3G 1A8, Canada, matthew.masapollo@mail.mcgill.ca), Athena Vouloumanos (New York Univ., New York, NY), and Lucie Ménard (Univ. of PQ at Montreal, Montreal, QC, Canada)

Recent research shows that pre-babbling infants can recognize infant-produced vowels as phonetically similar to adult and child vowel productions (Polka *et al.*, submitted), indicating that infants normalize for speaker size information. Yet little is known about whether infants encode information about speaker size in speech sounds and then use this information for source identification. Here, we investigate whether infants preferentially attend to smaller visual objects over larger visual objects when they hear infant speech sounds, and attend to larger visual objects over smaller visual objects when they hear adult speech sounds. We are currently testing 9-month-old infants using an intermodal matching procedure, in which they are presented with isolated vowel sounds synthesized to emulate productions by either adult female or infant speakers, along with side-by-side geometric shapes that differ in size (e.g., a large square and a small square). Preliminary analysis suggests that infants display greater mean proportion-looking times to the congruent shape-voice pairs, but additional data collection is ongoing. The implications of these findings for theories of infant speech perception will be discussed.

5aSC2. Interaction of memory and specificity in auditory repetition priming. Georgia Zellou and David Embick (Linguist, Univ. of Pennsylvania, 800 N. 48th St., #26, Philadelphia, PA 19139, gzellou@sas.upenn.edu)

This study examines the influences of abstract and episodic representations of words in auditory repetition priming. Two manipulations of prime and target items were employed, token-change and voice-change. First, either the target item consisted of a different token spoken by the same speaker as the prime (token-change), or the target item was a token spoken by a different speaker/different gender (voice-change) than the prime. Second, lapse between prime and target varied across three conditions: no-lapse (no intervening trials between prime and target), medium-lapse (exactly ten intervening trials), and long-lapse (exactly 20 intervening trials). Stimuli were presented using an auditory lexical decision task (with an equal number of English-like nonwords). The reaction time data reveal the greatest facilitation between token-change no-lapse prime-target pairs (256 ms), followed by voice-change no-lapse pairs (182 ms). Facilitation effects were comparable across both the medium-lapse and long-lapse conditions, and there was no advantage for token-change over voice-change items. We interpret these results in terms of current models of lexical access in which episodic representation are available only immediately after presentation, while abstract representations are active for a longer time.

5aSC3. The relationship between speech perception and production: Evidence from children with speech production errors. Kathryn Cabbage (Commun. Sci. and Disord., MGH Inst. of Health Professions, 79 13th St., Boston, MA 02129, klcabbage@yahoo.com) and Thomas Carrell (Special Education and Commun. Disord., Univ. of Nebraska-Lincoln, Lincoln, NE)

Children with speech sound disorders (SSD), for reasons that are not well understood, fail to produce age-appropriate speech production targets in isolation, words, and/or functional conversation. One line of research suggests these children exhibit underlying perceptual deficits for acoustic cues that distinguish similar phonemes, exhibiting particular deficits for those phonemes produced in error (e.g., /r/, /l/) (Monnin and Huntington, 1974; Shuster, 1998), although other studies have reported mixed results in children with SSD (Locke, 1980; Rvachew and Jamieson, 1989). In the current study, children with SSD and typically developing children were presented with a perceptual discrimination task involving a phoneme produced in error by the SSD children (i.e., /r/). Findings revealed two separate subsets of children with SSD. The first exhibited poor discrimination of /r/ while a second exhibited exceptionally good discrimination of the same contrast. Given that all children with SSD exhibited distorted /r/ productions, a follow-up acoustic analysis is being conducted to identify specific acoustic characteristics of the distorted /r/s in the two groups (i.e., the poor /r/ discriminators versus the excellent /r/ discriminators). The pattern of results relating the acoustical details of /r/ production in these two populations should contribute to a more refined model of speech sound disorders.

5aSC4. Effects of talker sex on phonetic convergence in shadowed speech. Jennifer Pardo, Hannah Gash, Adelya Urmanche, Alexa Decker, Keagan Francis, Jaclyn Wiener, and Sara Parker (Psych., Montclair State Univ., 1 Normal Ave., Montclair, NJ 07043, pardo@optonline.net)

Many studies have reported phonetic convergence during speech shadowing and conversational interaction, with highly variable results. Some of this variability is likely due to effects of talker sex on phonetic convergence. For example, some studies only use a single male model, with both male and female shadowers, while others focus on only male or female talkers. To date, few studies have provided a rigorous investigation of variability across male and female talkers in phonetic convergence. Of these, some studies have found that women converged more than men, while others report the opposite pattern. The current study examined phonetic convergence in a speech shadowing task with 48 talkers (24 female) who shadowed same sex or opposite sex models. The results were analyzed to reveal effects of talker sex and item type (mono- versus bisyllabic words) on measures of phonetic convergence. Because phonetic convergence varies according to the sex of the talker and the model, future studies should employ talker sets that are balanced with respect to talker sex.

5aSC5. Interdependent processing of speech and background noise. Angela Cooper (Linguist, Northwestern Univ., 2016 Sheridan Rd., Evanston, IL 60208, akcooper@u.northwestern.edu), Susanne Brouwer (Behavioural Sci., Utrecht Univ., Utrecht, Netherlands), and Ann R. Bradlow (Linguist, Northwestern Univ., Evanston, IL)

Speech processing can often take place in listening conditions that involve the mixing of speech and background noise. This study used a speeded classification paradigm to investigate whether background noise is perceptually integrated with indexical (Exp. 1) and phonetic (Exp. 2) dimensions of the speech signal. In each experiment, English listeners classified words along one of two dimensions: noise (pure tone vs. white noise) or a speech dimension, either gender (male vs. female in Exp. 1a), talker identity (Sue vs. Carol in Exp. 1b), or phoneme (/p/ vs. /b/ in Exp. 2), while ignoring the other dimension, which could be held constant (control), co-vary (correlated), or vary randomly (orthogonal). The results indicated that background noise was not completely segregated from speech, even when the two auditory streams were spectrally non-overlapping. Perceptual interference was asymmetric, whereby irrelevant indexical and phonetic variation slowed noise classification to a greater extent than the reverse. This suggests that while context-specific information (e.g., noise) and within-signal speech features are coupled together, they are unevenly weighted during this early stage in processing. This asymmetry may stem from the fact that speech features have greater salience and are thus more difficult to selectively ignore than environmental noise.

5aSC6. Speaking style adaptations across the lifespan. Rachael C. Gilbert, Cristabella Trimble-Quiz, Karen Johnson, and Rajka Smiljanic (Linguist, Univ. of Texas at Austin, 4812 Ave. H, Apt B, Austin, TX 78751, rachaelgilbert@gmail.com)

While intelligibility-enhancing speaking style modifications have been well researched in young adults, little is known about the development of speaking style adaptations across age. In particular, research on intelligibility-enhancing strategies in response to the environment (noise-adapted speech) and to the listener (clear speech) in children and older adult talkers is limited. In order to better understand how speech intelligibility varies with age, the present study examined conversational and clear speech sentences produced in quiet and in response to noise by 10 children (11–13 years old), 10 young adults (18–29 years old), and 10 older adults (60–84 years old). Twenty-two young adult listeners participated in word-recognition-in-noise and perceived age tasks. Results revealed that noise-adapted and clear speech modifications increased intelligibility for all talker groups compared to conversational speech produced in quiet, although children were marginally less intelligible. Perceived age strongly correlated with chronological age. Measures of F0, vowel space area, voice quality, spectral energy, and speaking rate were obtained for all talkers. Age-related differences were found for speaking rate, total energy in the 1-to-3 kHz region, and F0 mean. Further analyses will examine to what extent these acoustic-phonetic characteristics correlate with increases in intelligibility across speaking style adaptations and talker groups.

5aSC7. Talker's perspective modulates affective coherence in sentence comprehension. Hugo Quene, Anne R. Van Leeuwen, and Jos Van Berkum (Utrecht inst Linguist OTS, Utrecht Univ., Trans 10, Utrecht 3512JK, Netherlands, h.quene@uu.nl)

Does an audible frown or smile affect speech comprehension? Previous research suggests that a spoken word is recognized faster if its audible affect (frown or smile) matches its semantic valence. In the present study, listeners' task was to evaluate the valence of spoken affective sentences. Formants were raised or lowered using LPC to convey an audible smile or frown gesture co-produced with the stimulus speech. A crucial factor was the talker's perspective in the event being described verbally, in either first or third person. With first-person sentences, listeners may relate the talker's affective state (simulated by formant shift) to the valence of the utterance. For example, in "I have received a prize," a smiling articulation is congruent with the talker having experienced a happy event. However, with third-person sentences ("he has received a prize"), listeners cannot relate the talker's affective state to the described event. (In this example, the talker's affect can be empathic and positive, or envious and negative.) Listeners' response times confirm this hypothesized interaction: congruent utterances are processed faster than incongruent ones, but only for first-person sentences. When listeners evaluate spoken sentences, they combine audible affect, verbal content, as well as perspective, in a sophisticated manner.

5aSC8. Native listeners' sensitivity to foreign accent in short, slightly accented utterances: Non-native vowels. Hanyong Park (Linguist, Univ. of Wisconsin-Milwaukee, Curtin Hall 523, P.O. Box 413, 3243 N. Downer Ave., Milwaukee, WI 53211, park27@uwm.edu)

This study investigated whether native listeners can detect a foreign accent in short, slightly accented utterances. To answer this question, we examined 20 native listeners' sensitivity (d' values) to a foreign accent in a one-interval discrimination task (i.e., Yes-no design). Six L1 Korean learners of L2 English with high L2 proficiency along with six natives speakers of English produced the test materials consisting of three English vowels /ɑ/, /eI/, and /æ/ by using the delayed repetition technique. The listeners were asked to judge whether the speaker was a native or a non-native speaker of English for each speech sample played. Results indicated that most listeners detected a foreign accent from hearing the vowel stimuli. Furthermore, the listeners detected a foreign accent more often from the /æ/ stimuli than the /ɑ/ or the /eI/ stimuli. In line with previous L2 research, these results demonstrate that certain L2 segments are more difficult to learn than others. These results also suggest that listeners are sensitive to foreign accent and that they do not need much information (e.g., monosyllabic words) to detect a foreign accent, even in proficient L2 learners' productions.

5aSC9. Longer words or shorter words: Which one is more effective in distinguishing between self-identified gay and heterosexual male speakers? Erik C. Tracy (Psych., Univ. of North Carolina Pembroke, PO Box 1510, Pembroke, NC 28372, erik.tracy@uncp.edu)

Prior research demonstrated that listeners relied primarily on vowels to determine a male speaker's sexual orientation (Tracy and Satariano, 2011). Additionally, listeners became more confident in their sexual orientation judgments if the utterance included more vowels (Tracy, 2013). Since multisyllabic utterances also contain more consonants, are listeners' judgments based on the number of vowels or consonants in an utterance? To investigate this question, monosyllabic, bisyllabic, and trisyllabic words were selected. Within each group of words, the number of phones varied. For example, a trisyllabic word contained either seven phones ("division") or nine phones ("contribute"). Listeners became more confident of the speaker's sexual orientation if the word included more vowels. With respect to the monosyllabic and bisyllabic items, listeners were less confident if the number of consonants in the word increased. Ratings were more confident for "have" compared to "help". The results for the trisyllabic words differed. Listeners became more confident if the word contained more consonants. "Contribute" resulted in more confident ratings than "division". The results demonstrated that listeners were more confident in the speaker's sexual orientation if the word included more vowels, but under certain conditions, confidence ratings did not improve if the utterance included additional consonants.

5aSC10. Emergent tonogenesis in Afrikaans. Andries W. Coetzee, Patrice S. Beddor (Dept. of Linguist, Univ. of Michigan, 440 Lorch Hall, 611 Tappan Str, Ann Arbor, MI 48109, coetzee@umich.edu), and Daan P. Wissing (Ctr. for Text Technol., North-West Univ., PotchefstRm., South Africa)

Afrikaans is usually described as having a contrast between prevoiced and voiceless unaspirated plosives ([b]-[p], [t]-[d]). This study documents an ongoing change in the pre-vocalic realization of the contrast. Preliminary data from five speakers show that all speakers produce some phonologically voiced plosives without prevoicing, with frequency of devoicing ranging from 30% to 85% across speakers. This devoicing is nearly categorical: VOTs of devoiced plosives average 12 ms and those of phonologically voiceless plosives average 17 ms. (By comparison, VOTs of voiced plosives average around Ő130 ms.) The contrast appears to be preserved in the F0 contour of the following vowel, which is 50 to 100 Hz lower (depending on the speaker) after phonologically voiced than after phonologically voiceless plosives. The F0 difference continues through at least 70% of the vowel.

However, post-plosive F0 variation is not contingent on devoicing, e.g., F0 contours after phonological /b/ are the same regardless of whether the production is [p] or [b]. In these preliminary data, the magnitude of the F0 difference is linked to speaker age, with younger speakers showing a larger difference. Data from a larger group of speakers are being analyzed and will be presented.

5aSC11. Visualization of time-varying joint development of pitch and dynamics for speech emotion recognition. Chung Lee, Simon Lui (The Information Systems Technol. and Design Pillar, Singapore Univ. of Technol. and Design, 20 Dover Dr., Singapore 138682, Singapore, chung_lee@sutd.edu.sg), and Clifford So (School of Continuing and Professional Studies, Chinese Univ. of Hong Kong, Hong Kong, Hong Kong)

In this paper, we propose a new approach for visualizing the time-varying acoustic features for speech emotion recognition. Although the emotional state does not carry any linguistic information, it is a crucial factor that offers sentiment feedback to the listener. We propose to extract the two most prevalent acoustic features: pitch and dynamics, to identify the speech emotion of the speaker. We represent the time-varying pitch and dynamics as a trajectory in a two-dimensional feature space. Multiple trajectories are then segmented and clustered into signature patterns. This technique was successful in identifying and retargeting expressive musical performance styles. In evaluation, we use the German emotion language database. The database was created with ten professional actors (five males and five females) of ten emotionally unbiased sentences performed in six target emotions (Angry, Happy, Fear, Boredom, Sad, and Disgust). Results showed that the speech samples from the same actor of the same sentence but different emotions have dramatically different trajectory patterns. On the other hand, obvious common patterns were found among low valence emotions like Boredom and Sadness. The current study also opens future research opportunities for applying advanced pattern recognition techniques (e.g., Support Vector Machine and Neural Network) for better emotion identification.

5aSC12. Vowel discrimination at high fundamental frequencies in real speech. Daniel Friedrichs (Phonet. Lab., Univ. of Zurich, Plattenstrasse 54, Zurich 8004, Switzerland, daniel.friedrichs@uzh.ch), Dieter Maurer, Heidy Suter (Inst. for the Performing Arts and Film, Zurich Univ. of the Arts, Zurich, Switzerland), and Volker Dellwo (Phonet. Lab., Univ. of Zurich, Zurich, Switzerland)

Previous research showed that, in singing, vowel qualities of isolated vowel sounds can be discriminated up to a fundamental frequency (F0) of about 500 Hz. However, indications are reported in literature for vowel discrimination on F0 > 500 Hz for singing (raised larynx condition, CVC context) as well as for speech-like sounds. In this study, we tested vowel discrimination at a high F0 in speech using minimal pairs build from eight long German vowels. Words were produced in speech mode at F0 of about 650 Hz by two female speakers. For all samples except the words including /a/ and /ɛ/, F0 exceeded F1 values as given in vowel statistics for Standard German. In a listening test, stimuli were played back in random order to 14 listeners (7f, 7m) for identification. The results showed that vowel discrimination can be preserved at such high fundamental frequencies. This could mean that, for our speakers and the high fundamental frequency examined, (1) source-filter-characteristics were effective up to 650 Hz, or (2) transitions played a crucial role, or (3) other spectral characteristics than formants have to be taken into account in order to explain these results.

5aSC13. Perceptual importance of time-domain features of the voice source. Marc Garellek (Linguist, Univ. of California, San Diego, La Jolla, CA), Gang Chen (Elec. Eng., Univ. of California, Los Angeles, Los Angeles, CA), Bruce R. Gerratt (Head and Neck Surgery, Univ. of California, Los Angeles, Los Angeles, CA), Abeer Alwan (Elec. Eng., Univ. of California, Los Angeles, Los Angeles, CA), and Jody E. Kreiman (Head and Neck Surgery, Univ. of California, Los Angeles, 31-24 Rehab Ctr., 1000 Veteran Ave., Los Angeles, CA 90403, jkreiman@ucla.edu)

Our previous study examined the perceptual adequacy of different source models. We found that perceived similarity between modeled and natural voice samples was best predicted (in the time dimension) by the match between waveforms at the negative peak of the flow derivative (R^2 = 0.34). The extent of fit during the opening phase of the source pulses added only 2% to perceived match. However, in that study model, fitting was unweighted, and results might differ if another approach were used. In this study, we constrained the models to fit the negative peak of the flow derivative precisely. We fit 6 different source models to 40 natural voice sources, and then generated synthetic copies of the voices using each modeled source pulse, with all other synthesizer parameters held constant. We then conducted a visual sort-and-rate task in which listeners assessed the extent of perceived match between the original natural voice samples and each copy. Discussion will focus on the specific strengths and weaknesses of each modeling approach for characterizing differences in vocal quality, and on the importance of matches to specific time-domain events versus spectral features in determining voice quality. [Work supported by NIH/NIDCD grant DC01797 and NSF grant IIS-1018863.]

5aSC14. Using acoustic string-edit distance to evaluate US pronunciation variation. Clelia R. LaMonica (Dept. of English, Stockholm Univ., Stockholm 106 91, Sweden, clelia.lamonica@english.su.se)

String-edit distances, such as the Levenstien distance, have been used in perceptual-linguistic studies to compare differences among language varieties, based on changes that occur between two speech samples (Gooskens and Heeringa, 2004; Heeringa *et al*., 2006; Nerbonne *et al*., 2008). However, this method relies mainly on phonetic transcriptions rather than actual speech data. This work illustrates how acoustic measurements are taken from speech samples from across the United States, and a distance measurement between them is derived for use in further perceptual comparisons (such as perceived distance from standard, intelligibility, appeal, and identification). The speech samples consist of six sentences from various regions of the United States, each sentence containing phonological features that may be marked as perceptually significant for dialect identification (Clopper 2011, Labov *et al.* 2005, Thomas 2001). The methodology for assessing phonetic distance between two regional varieties is addressed, in particular, by using the Euclidean distance between normalized Bark formants of phonological features within the samples. The benefits and potential disadvantages to using acoustic data vs. transcribed data are addressed as well.

5aSC15. Mapping accent similarity and speech in noise intelligibility for British English accents. Paul Iverson, Melanie Pinet, and Bronwen G. Evans (Speech Hearing and Phonetic Sci., Univ. College London, 2 Wakefield St., London WC1N 1PF, United Kingdom, p.iverson@ucl.ac.uk)

Previous work has suggested that the recognition of speech in noise is affected by the accent similarity of the speaker and listener, as well as by the familiarity of the speaker's accent. The present study investigated this further by constructing multidimensional accent maps for British English speakers, as well as a small group of general American speakers, and examining how intelligibility within this space varied for listeners with different British English accents. The preliminary results suggest that speakers who have a central position in the accent space are most intelligible in noise, and that speakers with standard accents tend to occupy this central position. This implies that some aspects of speech intelligibility may be explained by the prototypicality of speakers within the broader accent space.

5aSC16. Lexically mediated perceptual learning generalizes to new word positions. Alexis R. Johns and James S. Magnuson (Univ. of Connecticut, 406 Babbidge Rd., Unit 1020, Storrs, CT 06269, alexis.johns@uconn.edu)

Acoustic properties of words vary based on idiolects of different speakers, yet listeners appear to understand this varying speech input effortlessly. How do listeners adapt to acoustic variation across speakers? Previous research has shown that listeners can implicitly shift phonetic category boundaries in a speaker-specific fashion during a two-step paradigm involving lexical decision on critical words with a word-medial ambiguous phoneme, followed by a phonetic categorization task of that ambiguous phoneme (see Kraljic and Samuel, 2005). Other studies report that this boundary shift does not generalize to other speakers (Kraljic *et al*., 2008).

However, for a given speaker, it does generalize to different positions within a word, for instance, from word-medial to word-initial (Jesse and McQueen, 2011). Importantly, these generalization tests used identical ambiguous phonemes in each position, rather than context-appropriate tokens, allowing the possibility of token-specific learning rather than true generalization. The current research extends these findings by providing distinct, position-appropriate tokens of the ambiguous phoneme during training and test items, and shows that the generalization to new word position is retained. Thus, generalization within a speaker is not token-specific, but instead exhibits some level of abstraction.

5aSC17. Perception of second language phoneme masked by first- or second-language speech in 20–60 years old listeners. Rieko Kubo, Masato Akagi (JAIST, 1-1 Asahidai, Nomi, Ishikawa 923-1292, Japan, rkubo@jaist.ac.jp), and Reiko Akahane-Yamada (ATR-IRC, Seika, Japan)

In previous research, we conducted a perceptual training on American English /r/-/l/ for Japanese adults in various age groups and found that; although the training improved identification of these phonemes in all age groups, the improvement decreased along the ages. Analyses suggested that perceptual cues of second-language (L2) phoneme perception differ across age groups. This paper investigated the decrease by focusing on perceptual cues. Listening tests were performed to assess intelligibility of Japanese words and English minimal-pairs masked by English or Japanese speech. Theoretically, in case L2 targets have phonologically non-equivalent in listener's first-language (L1) orthography, L1-relevant cues can be used separating the targets. In case L2 targets have phonologically equivalent in listener's L1 orthography, L2-relevant acoustic cues should be taken into perceptual cues. Masker having identical language to the perceptual cues may interfere identifying the targets. In the results of English targets which were supposed to be phonologically equivalent in Japanese orthography, young-adult Japanese showed higher effect from English masker than that of Japanese masker; conversely, older-adult Japanese showed higher effect from Japanese masker. The results suggested that young adults more focused on L2-relevant acoustic cues than elderly did. This difference may lead to the decrease in training effect.

5aSC18. Effect of aging on auditory processing: Relationship between speech perception and auditory brainstem responses. Su-Hyun Jin and Won So (Commun. Sci. and Disord., Univ. of Texas at Austin, 1 University Station, A1100, Austin, TX 78712, shjin@utexas.edu)

The purpose of the study is to evaluate the auditory processing abilities of older (60–75 years old) and younger (18–30 years old) listeners with normal hearing up to 4000 Hz. We are going to measure series of behavioral and physiological responses of these two listener groups. Behavioral measures include listener's hearing sensitivity, speech recognition in various listening conditions using different types of noise, and temporal resolution processing complex time-varying signals. For a physiological measure, speech-evoked ABRs, neural responses of subcortical auditory nuclei will be recorded. This quick and non-invasive technique allows us to measure neural responses to fast time-varying speech stimuli. We already synthesized speech stimuli, short syllable like /da/, /ba/, /ta/, /pa/ that can be used to evoke the ABR. It is hypothesized that compared to younger group, older listeners might show a loss of temporal precision results in abnormal ABRs such as subcortical timing delays and decreases in response consistency and magnitude. In addition, listeners with abnormal ABRs might show poorer performance in speech recognition in noise as well as reduced temporal resolution processing. That is, there would be significant correlation between behavioral and physiological measures.

5aSC19. Listen to your mother: Highly familiar voices facilitate perceptual segregation. Ingrid Johnsrude, Elizabeth Casey (Dept. of Psych., Queen's Univ., 62 Arch St., Kingston, ON K7L 3N6, Canada, ij4@queensu.ca), and Robert P. Carlyon (Cognition and Brain Sci. Unit, Medical Res. Council, Cambridge, United Kingdom)

We studied the effect of voice familiarity on the ability to segregate one voice from a competing speaker. Specifically, we examine the utility of arguably the most familiar voice of all—the mother's voice—in facilitating segregation, and compared it to the effect of a voice that listeners had been familiarized with in the laboratory. We tested 19 older adolescents (still living at home) on a version of the coordinate-response-measure procedure (CRM; Bolia *et al.*, 2001), with mixtures of two voices, at three signal-to-noise ratios (Ő3 dB, 0 dB, + 3 dB). Performance was better when the mother's voice was the target, compared both to novel and lab-familiar targets. At the most disadvantageous target-to-masker ratio (Ő3 dB), listeners were also better able to ignore their mother's voice so as to comprehend a stranger's voice more effectively, demonstrating that extremely familiar voice information facilitates segregation. This pattern of results is similar to that observed with older people (aged 44–59) when their spouse's voice was present in a two-voice CRM mixture (Johnsrude *et al.*, 2013). The new results demonstrate the importance of long-term (rather than short-term) familiarity and show that it aids sound segregation for adolescents as well as older adults.

5aSC20. Audio/visual correlates of a vowel near-merger in the Bay Area. Keith Johnson and Sarah Bakst (Linguist Dept., Univ. of California Berkeley, Berkeley, CA 94720, keithjohnson@berkeley.edu)

This study is a part of a research program aiming to learn the extent to which visual phonetic information may be involved in the maintenance of an acoustically weak or marginal contrast. The "caught/cot" contrast is one such contrast which has been merged in much of the western United States. We seek to document both the presence of the contrast in the SF Bay area and the phonetic realization of it. We will be reporting on audio/visual recordings from 33 native English speakers from the Bay Area (14 from San Francisco). Participants read a brief passage and a set of sentences that targeted these vowels, and then completed a commutation task where speakers tried to identify their own productions of the words "cot" and "caught." We found that three of the speakers reliably made the caught/cot contrast, and phonetic analysis confirms that regardless of whether speakers perceive a distinction, the vowels may differ acoustically or on degree of visual lip rounding, depending on the context. The two vowels tend to be most distinct in the single-word context and the least in the passage context. [Funded by the NSF.]

5aSC21. Acoustic cue weighting in perception and production of English and Spanish. Jessamyn L. Schertz, Natasha Warner, and Andrew J. Lotto (Univ. of Arizona, Douglass 200, Tucson, AZ 85721, jschertz@email.arizona.edu)

We present data quantifying the use of multiple acoustic dimensions in the perception and production of word-initial stops in Spanish and English in bilinguals and monolinguals, with a focus on the stability of individuals' cue weighting strategies across languages and modalities. Early Spanish-English bilinguals who use both languages on a regular basis, along with control groups of monolingual Spanish and English speakers, categorized stop-initial CV syllables covarying in Voice Onset Time (VOT), fundamental frequency at vowel onset (f0), first formant onset frequency (F1), and stop closure duration. Individual perceptual cue weights were calculated via logistic regression, and these weights were compared with the same speakers' use of the corresponding acoustic dimensions in their productions of the stop contrast. Preliminary results suggest that bilingual listeners' perceptual cue weighting strategies correspond across Spanish and English; that is, individuals who weight a given cue more heavily than average in English also weight that cue heavily in Spanish. However, in line with previous work, correspondences between a given individual's use of the acoustic dimensions across perception and production are more elusive.

5aSC22. Effects of auditory and visual variability on word learning in children. Kelly Casey, Shayna Marmon, Emily Butterworth, Elana Katz, Kristin A. Vasil-Dilaj, and Rachel M. Theodore (Univ. of Connecticut, 850 Bolton Rd., Unit #1085, Storrs, CT 06269, rachel.theodore@uconn.edu)

One major feat in development is acquiring vocabulary for nouns, a process that necessitates learning a systematic pairing between an auditory and a visual stimulus. This is computationally complex because the acoustic signal produced for a given noun, such as "dog," varies considerably as do objects in the environment that are given this label. Studies that have separately examined auditory and visual variability suggest that variability in either domain can facilitate word learning. The goal of the current work is to examine simultaneous contributions of auditory and visual variability on word learning. Typically developing infants (15–20 months of age) participate in nine weekly sessions. Each session entails exposure to nouns in a naturalistic play environment where children manipulate visual exemplars while listening to auditory exemplars. Auditory exemplars consist of productions by either ten talkers or by a single talker. Visual exemplars consist of objects that are highly variable or minimally variable in appearance. We measure learning for training exemplars, generalization to novel exemplars, and vocabulary size. Our research questions are (1) does variability in the auditory domain alone facilitate word learning as has been shown for visual variability, (2) to what degree does simultaneous variability in both modalities influences noun learning.

5aSC23. An auditory test battery for the assessment of executive functions. Blas Espinoza-Varas (Commun. Sci. & Disord., OU Health Sci. Ctr., 1200 N. Stonewall Ave., Oklahoma City, OK 73117-1215, blas-espinoza-varas@ouhsc.edu), Kai Ding, and Sudha Lakhwani (Biostatistics & Epidemiology, OU Health Sci. Ctr., Oklahoma City, OK)

Recent studies have examined the potential role of executive functions (EFs) on the speech communication ability of cochlear-implant, hearing-aid users, and other patients with disordered speech, language, or hearing; in many of the tests used to measure EFs, the stimuli are visual rather than auditory. We developed an Auditory Executive Function (AEF) test battery in which the stimuli consist of word commands (e.g., "quit," "stop") spoken either in stern or in lenient voice tone. The battery measures the speed and accuracy of voice-tone classification in four conditions: (1) in the absence of information or response conflict (baseline); (2) while attempting to inhibit potent but inappropriate responses prompted by conflicting ear-laterality information (inhibitory control); (3) while having to switch between incompatible response-mapping rules from trial to trial (cognitive flexibility); and (4) while having to monitor and update the word presented in the current trial and remember the word presented 2–3 trials prior to the current one (working memory). Relative to baseline, classification accuracy decreases with conflicting laterality information and more so with trial-to-trial switching of response-mapping rules, but only the latter condition decreased the classification speed. The AEF test battery is portable and fully automated. [Research supported by the Alcohol Beverage Medical Research Foundation.]

5aSC24. Perception of speaker sex in re-synthesized children's voices. Peter F. Assmann, Michelle R. Kapolowicz, David A. Massey (School of Behavioral and Brain Sci., Univ. of Texas at Dallas, MS GR 41, Box 830688, Richardson, TX 75075, assmann@utdallas.edu), Santiago Barreda (Dept. of Physiol., Univ. of Arizona, Tucson, AZ), and Terrance M. Nearey (Dept. of Linguist, Univ. of AB, Edmonton, AB, Canada)

Recent studies have shown that fundamental frequency (F0) and average formant frequencies (FF) provide important cues for the perception of speaker sex. Experiments on vocoded adult voices have indicated that upward scaling of F0 and FFs increases the probability that a voice will be perceived as female while downward scaling increases the probability that the voice will be perceived as male. The present study extends these manipulations to children's voices. Separate groups of adult listeners heard vocoded /hVd/ syllables spoken by five boys and five girls from 14 age groups (5–18 years) in four synthesis conditions using the STRAIGHT vocoder. These conditions involved swapping F0 and/or FFs to the opposite-sex average within each age group. Compared to the synthesized, unswapped originals, both the F0-swapped condition and the FF-swapped condition resulted in lower sex recognition accuracy for the older females but relatively smaller effects for males. The combined F0 + FF swapped condition produced the largest drop in performance for both sexes, consistent with findings indicating that a change in F0 or FFs alone is generally insufficient to produce a compelling conversion of speaker sex in adults. [Hillenbrand and Clark, Attention Percep. Psychophys. **71**(5), 1150–1166 (2009).]

5aSC25. Voice quality variation across gender identities. Kara Becker, Sameer ud Dowla Khan, and Lal Zimman (Linguist, Reed College, 3203 SE Woodstock Bvld., Portland, OR 97202, sameeruddowlakhan@gmail.com)

While several studies of the phonetics of American English make reference to general features of "women's speech" and "men's speech," no large-scale acoustic or articulatory study has established the actual range and diversity of phonetic variation along gender identities, encompassing different sexual orientations, regional backgrounds, and socioeconomic classes. The current study explores the acoustic and articulatory phonetic features related to gender, particularly focusing on voice quality, e.g., creaky and breathy phonation. Subjects identifying with a range of gender and other demographic identities were audio recorded producing the Rainbow Passage as well as wordlists targeting variation in vowel quality. Simultaneously, electroglottographic (EGG) readings were taken and analyzed for voice quality. Subjects were then asked to rate recordings of other people's voices to identify the personal characteristics associated with the acoustic reflexes of phonation; in the final task, subjects were also explicitly asked about their language ideologies as they relate to gender. Thus, the current study explores the relation between gender identity and phonetic features, measured acoustically, articulatorily, and perceptually. This work is currently underway and preliminary results are being compiled at this time.

5aSC26. The role of acoustics in processing syntactically ambiguous sentences. Victoria Sharpe (Honors College, Univ. of South Carolina, 104 Mount Elon Church Rd., Hopkins, SC 29061, sharpevp@email.sc.edu), Dirk-Bart den Ouden, and Daniel Fogerty (Dept. of Commun. Sci. and Disord., Univ. of South Carolina, Columbia, SC)

While natural speech prosody facilitates sentence processing, unnatural prosody is expected to decrease speed and accuracy in resolving syntactic ambiguities. This study investigated how, and to what extent, fundamental frequency (F0) and temporal envelope (E) contribute to sentence processing. Signal processing methods degraded either F0 or E in 120 garden-path sentences with a temporarily ambiguous structure (e.g., While the man hunted the deer ran into the woods). Participants listened to natural and acoustically modified garden-paths and, for each, answered a comprehension question and repeated the sentence. Results demonstrated that degrading E consistently affected sentence comprehension, with a different effect observed for degrading F0. Interestingly, the effect of acoustic degradation interacted with effects of verb type and sentence plausibility. In the E-modified condition, sentences with Reflexive Absolute Transitive verbs in which the erroneous interpretation was plausible (e.g., While Mary bathed the baby fell in the tub—*Mary bathed the baby) resulted in productions that were relatively shorter in duration and that reconstructed the original sentence prosody. These findings suggest E plays a greater role, compared to F0, in disambiguating syntax. However, use of prosody by listeners may vary based on context, suggesting prosodic information can interact with cognitive processing load.

5aSC27. Acoustic separability of Mandarin tonal categories in careful and conversational speech. Daniel Brenner (Linguist, Univ. of Arizona, 814 E 9th St. Apt. 14, Tucson, AZ 85719-5322, dbrenner@email.arizona.edu)

Previous work has found that tonal categories in conversation are realized with mostly much weaker acoustic cues than in careful speech. For example, pitch curves become flatter (less curved), more level (shallower slopes), and more centralized (closer to the speaker's mean speaking pitch) in conversation on average relative to careful speech, and also more variable. At the same time, a few cues appear to change direction (e.g., the pitch range of tone 4 is much larger than that of tone 3 in careful speech, but is far shorter in conversation), or become acoustically salient to categories in conversation only (pilot results suggest that tone 3 is especially short in duration in conversation). The present project incorporates tonal context, and bigram syllable- and tone- frequencies into a model of the differentiability of second-syllable tonal categories in two-syllable Mandarin words in conversation. This model approximates the extent to which tonal category can be determined from available acoustic cues in conversation while accounting for co-occurrence frequencies.

5aSC28. Production and perception of the pre-lateral, non-low, back vowel merger in northeast Ohio. Lacey R. Arnold (English, North Carolina State Univ., 1214 Carlton Ave., Raleigh, NC 27606, lrarnold@ncsu.edu)

This study examines the status of the pool/pull, pole/pull, and pool/pull/pole mergers in the Youngstown, Ohio area, as well as various factors that may influence perception of these mergers. The participants include 16 natives of the Youngstown, Ohio area, ranging from ages 9–66, and making up two separate families; one individual belonged to neither. The findings suggest that all three mergers can be found in the Youngstown area, though each is undergoing a different process. Results of the perception task suggest that individuals can use additional cues, such as vowel duration, to aid in identification of words containing merged phonemes. Findings also suggest that homophonous words uttered by merged speakers in the context of a carrier sentence may contain additional information that listeners are able to access and utilize for word identification. Lastly, the effects of familiarity with the speaker and merged/distinct production on word identification reveal no definitive results, though the study's findings suggest further investigation of these factors would be a worthwhile endeavor.

5aSC29. Perceptual tonal spaces in whispered speech. Grace Kuo (Linguist, Macalester College, 3125 Campbell Hall, Los Angeles, CA 90095, gracekuo@humnet.ucla.edu)

Whispered speech is made when no vocal fold vibration occurs. However, pitch from whispers can still be perceived for some listeners because it corresponds to the change in F1 and F2. Listeners rely on the laryngeal movements, which not only associate with pitch change but also result in the change of the tongue position as well as the shape of the oral cavity. This study examines the acoustical-perceptual relationships in the identification of pitch during whispered tone production. Listeners (English and Taiwanese) rate the similarity of fifty AX tonal pairs (Taiwanese [ka͡ʊ] and [ç] in modal voice vs. whispered voice). Multidimensional scaling (MDS) analysis is used to map English listeners' and Taiwanese listeners' perceptual tonal spaces regarding modal speech and whispered speech stimuli. The correlations between the similarity ratings, the reciprocal of their reaction time, and the selected acoustic measures are examined.

5aSC30. Dynamic pitch and pitch range interact in distortions of perceived duration of American English speech tokens. Alejna Brugos and Jonathan Barnes (Boston Univ., 14 Asylum St., Mendon, MA 01756, abrugos@bu.edu)

Previous research showed that pitch factors can distort perceived duration: tokens with dynamic or higher f0 tend to be perceived as longer than comparable level-f0 or lower-f0 tokens, and silent intervals bounded by tokens of widely differing pitch are heard as longer than those bounded by tokens closer in pitch (the kappa effect). Fourteen subjects were asked to judge which of two exemplars of a spoken word sounded longer. All tokens were created from the same base file with manipulations of objective duration, f0 contour (plateaux vs. rises of different slopes) and pitch range. Results show that pitch range relation between the two exemplars was a stronger predictor of perceived duration distortion than f0 contour. In addition to previously demonstrated effects of f0 height (Yu, 2010), greater f0 discontinuity between tokens increases the likelihood that the first token of a pair will be judged as longer, suggesting that some previous findings showing the effects of dynamic pitch on perceived duration may actually be magnified by the kappa effect. Listeners may be responding to perceived prosodic distance that integrates information from timing (filled and silent intervals) and pitch (pitch slope and pitch jumps across silent intervals).

5aSC31. Indexical variation affects semantic spread. Ed King and Meghan Sumner (Linguist, Stanford Univ., 450 Serra Mall, Bldg. 460, Rm. 127, Stanford, CA 94305, etking@stanford.edu)

The role of indexical variation in spoken word recognition is constrained to acoustically rich lexical representations. Theoretically, lexical activation depends on indexical variation, but subsequent processes like associative semantic spread depend on activation strength, not indexical variation. Social psychological theories view indexical variation as integral to online processes such as persona construal. Therefore, information gleaned from indexical variation might pervade spoken word recognition more broadly. We investigate the effects of indexical variation on semantic activation in word-association and semantic-priming paradigms. Across three studies, we show that top associate responses depend on the voice of the probe word ("space" in man's voice: time; woman's voice: star; child's voice: planet). Voice also affects response frequency distributions: the man's voice receives a wider variety of weaker responses, while the woman's and child's voices receive fewer, stronger, responses. We also find that semantic priming varies as a function of voice-specific word association strength: priming is stronger to strong voice-specific associates (woman: space-star) than to weak associates (woman: space-time). We argue that indexical variation affects spoken word recognition beyond an episodic lexicon and provide an account capturing effects of learned associations between acoustic patterns and linguistic and social features in spoken language processing.

5aSC32. Effects of listener characteristics on foreign-accentedness rating of a non-standard English dialect. Andrea Morales and Natasha Warner (Linguist, The Univ. of Arizona, 5242 S Hampton Roads Dr., Tucson, AZ 85756, andreamorales@email.arizona.edu)

This project analyzes what characteristics of listeners affect whether they perceive Chicano English as foreign-accented English. Many Americans assume Chicano English (CE) is non-native English spoken by native Spanish speakers, but CE is often spoken as a native dialect of English. CE is a very common dialect in Tucson, Arizona, and this project examines the correlation between listeners' ethnicity, familiarity with Hispanic people, and political stance on immigration, and their perception of CE as foreign-accented. Stimuli are sentences read by CE and other Tucson speakers that contain phonetic environments where CE has features that distinguish it from Standard American English (SAE) . The listener population is Southern Arizonans of various ethnicities with varying degrees of exposure to CE and Spanish. The experiment uses a Foreign Accentedness Rating (FAR) task, as well as classification of stimuli as spoken by a Hispanic vs. Anglo speaker and background questions on listeners' language background and political opinions. Highly accurate identification of ethnicity is predicted, as well as correlations between some measures of the listeners' background and strength of FAR rating of CE speakers. Conclusions involve the effect of long-term exposure to a local dialect and sociolinguistic status on perceived degree of foreign accent.

Session 5aSP

Signal Processing in Acoustics: Signal Processing Models for Sound Production and Perception

Mark D. Skowronski, Chair
Communicative Sciences and Disorders, Michigan State Univ., Okemos, MI 48864

Contributed Papers

8:30

5aSP1. Distribution and propagation of sound sources in the vocal tract. Liran Oren, Sid Khosla, and Ephraim Gutmark (Univ. of Cincinati, PO Box 670528, Cincinnati, OH 45267, orenl@ucmail.uc.edu)

In the current study, an array of 128 microphones is strategically positioned outside a vocal tract model that is placed above an excised canine larynx. Acoustic holography technique is then used to identify the distribution of sound in the vocal tract by using simultaneous measurements from the microphone array. The results show that with no downstream constriction (i.e., open mouth) the energy coming from the higher harmonics is concentrated near the vibrating folds. When downstream constriction is added (i.e., partially closed mouth), the energy in the higher harmonics is shifted further downstream toward the mouth. These results suggest that acoustic holography can be used as a new tool to assess how sound sources propagate in the vocal tract during normal speech and consequently determine the severity of certain speech disorders such as hypernasality and nasal emission.

8:45

5aSP2. Accuracy of six techniques for measuring formants in isolated words. Christine H. Shadle, Hosung Nam (Haskins Labs., 300 George St., New Haven, CT 06511, shadle@haskins.yale.edu), and D. H. Whalen (Speech-Language-Hearing Sci., City Univ. of New York, New York, NY)

In previous work comparing formant measurements of synthetic vowels, manual measurements and four of the six algorithms tested were shown to have large errors and be biased by fundamental frequency (F0). In this study, natural speech recorded by five speakers (2 female, 3 male) was used to provide a more realistic test. The context /hVd/ was used with vowels V = {/i, ae, u, a/}, in which formant frequencies could be expected to be relatively constant; declarative intonation was elicited so that F0 would be falling throughout the vowel, eliciting evidence of F0 bias. Formants were estimated throughout using LPC Burg, LP closed-phase covariance, Weighted Linear Prediction with Attenuated Main Excitation (WLP-AME) [Alku *et al.*, JASA **134**(2), 1295–1313 (2013)], AVG, CEPS, and reassigned spectrogram with pruning (RS). The LPC methods show evidence of F0 bias, with WLP-AME having the smallest errors (Alku *et al.* compared it to many LPC variants, including DAP); RS has the smallest errors overall, but is not fully automatic. Accuracy during vocal fry and for low vs. high F0s will be discussed. As with synthetic speech, WLP-AME can be recommended as the analysis of choice until RS becomes more automatic and less subjective. [Work supported by NIH.]

9:00

5aSP3. Auditory-inspired pitch extraction using a synchrony capture filterbank for speech signals. Kumaresan Ramdas, Vijay Kumar Peddinti (Dept. of Elec., Comput. & Biomedical Eng., Univ. of Rhode Island, 4 East Alumni Ave., Kingston, RI 02881, kumar@ele.uri.edu), and Peter Cariani (Hearing Res. Ctr. & Dept. of Biomedical Eng., Boston Univ., Boston, MA)

The question of how harmonic sounds produce strong, low pitches at their fundamental frequencies, F0s, has been of theoretical and practical interest to scientists and engineers for many decades. Currently the best auditory models for F0 pitch, [e.g., Meddis and Hewitt, J. Acoust. Soc. Am. **89**(6), 2866–2894 (1991)] are based on bandpass filtering (cochlear mechanics), half-wave rectification and low-pass filtering (haircell transduction and synaptic transmission), channel autocorrelations (all-order interspike interval statistics) aggregated into a summary autocorrelation, and an analysis that determines the most prevalent interspike intervals. As a possible alternative to autocorrelation computations, we propose an alternative model that uses an adaptive Synchrony Capture Filterbank (SCFB) in which groups of filter channels in a spectral neighborhood are driven exclusively (captured) by dominant frequency components that are closest to them. The channel outputs (for frequencies below 1500 Hz) are then adaptively phase aligned with respect to a common time reference to compute a Summary Phase Aligned Function (SPAF), aggregated across all channels, from which F0 is easily extracted.

9:15

5aSP4. Acoustic characteristics of the Lombard effect from talkers with Parkinson's disease. Rahul Shrivastav (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI), Mark D. Skowronski (Communicative Sci. and Disord., Michigan State Univ., 2355 Club Meridian Dr., Apt. B02, Okemos, MI 48864, markskow@hotmail.com), Lisa M. Kopf, and Brad Rakerd (Communicative Sci. and Disord., Michigan State Univ., East Lansing, MI)

Speech production in noise elicits the Lombard effect, characterized by an increase in vocal effort. In talkers with Parkinson's disease (PD), noisy environments are a stressor to speech production. Recently, the intelligibility of speech from PD talkers speaking in noisy (additive multi-talker babble) and reverberant environments was shown to be more sensitive to both factors than speech from healthy talkers [Kopf *et al.*, ASHA **7387** (2013)]. The current study investigated the acoustic properties of speech and how they change in noise and reverberation for PD talkers compared to healthy talkers. Healthy talkers increased their level and fundamental frequency in additive noisy environments and, to a lesser degree, in reverberant environments. PD talkers also changed their level and fundamental frequency but to a more limited extent compared to healthy talkers. Additional changes in speech production (speaking rate, articulation range, and voice quality) were observed in individual talkers from both groups.

9:30

5aSP5. The cepstral peak: A theoretic analysis and implementation comparison of a popular voice measure. Mark D. Skowronski, Rahul Shrivastav, and Eric J. Hunter (Communicative Sci. and Disord., Michigan State Univ., 1026 Red Cedar Rd., Rm. 216, East Lansing, MI 48824, markskow@msu.edu)

Cepstral analysis of speech has traditionally provided two pieces of information: (1) the cepstral lag, which is a measure of fundamental frequency, and (2) the cepstral peak, which is a measure of the degree of periodicity. Cepstral analysis conveniently separates the effects of the vocal tract from the acoustic source without explicit vocal tract estimation. The cepstral peak of speech is a popular diagnostic tool for voice disorders and vocal treatment assessment, yet the expected value of the cepstral peak has not received sufficient theoretic treatment. Discrete-time analysis of the

cepstrum of a periodic pulse train revealed that the cepstral peak is 1/2 for all pulse trains with integer fundamental period T samples. For the more general case of a non-integer period, a pulse train formed with sinc functions produces a cepstral peak of $1/2+\varepsilon$ where the magnitude of ε scales with 1/T. The cepstral peak of various test signals was compared to cepstral peak prominence [Hillenbrand *et al*., J. Speech Hear. Res. **37**, 769–778 (1994)], and accuracy was improved by (1) zero-padding the log spectrum before inverse Fourier transformation, which provides cepstral interpolation, and (2) limiting spectral nulls, which trades off estimate variance and bias.

9:45

5aSP6. Objective measures of blind source separation and the intelligibility of separated speech. Richard Goldhor, Keith Gilbert (Speech Technol. & Appl. Res. Corp., 54 Middlesex Turnpike, Entrance D, Bedford, MA 01730, rgoldhor@sprynet.com), Suzanne Boyce, and Sarah Hamilton (Dept. of Commun. Sci. and Disord., Univ. of Cincinnati, Cincinnati, OH)

When multiple acoustic sources are present in an environment monitored by multiple microphones, each microphone's response signal is typically a weighted and delayed mixture of the active source signals. Subject to important constraints, these mixtures can be processed using Blind Source Separation (BSS) methods to construct output signals representative of each of the "hidden" acoustic sources in isolation. An important objective measure of the quality of a given "separation solution" is the amount of mutual information remaining in the BSS output channels: the less mutual information across channels, the better the separation. Often, the acoustic signals of interest in a complex acoustic environment are speech sources, and the measure of separation quality that really matters to listeners is speech intelligibility—a subjective (perceptual) measure that cannot be directly computed by a signal processing system. Thus, it would be useful to know how well objective measures of separation quality, such as mutual information, correlate with intelligibility. We present results of perceptual and objective tests exploring the relationship between objective separation metrics and word intelligibility as estimated by listeners' responses in word identification tasks, for target-word-in-carrier-phrase utterances recorded in an acoustic environment with masking speech babble and broadband noise sources.

10:00–10:15 Break

10:15

5aSP7. Toward a scalable, binaural hearing device: Acoustics, interaural processing, and scene decomposition. Birger Kollmeier, Marko Hiipakka, Giso Grimm, Tobias Neher, Tobias de Taillez (Cluster of Excellence Hearing4all & Medizinische Physik, Universität Oldenburg, Oldenburg D-26111, Germany, birger.kollmeier@uni-oldenburg.de), Jens Schröder (Project Group Hearing, Speech and Audio Technol., Fraunhofer IDMT, Oldenburg, Germany), Jörn Anemüller, and Volker Hohmann (Cluster of Excellence Hearing4all & Medizinische Physik, Universität Oldenburg, Oldenburg, Germany)

One of the aims of the focused research group "Individualized hearing acoustics" are elements of an assistive listening device that both fits the requirements of near-to-normal listeners (i.e., providing benefit in noisy situations or other daily life acoustical challenges) and, in addition, can be scaled up to a complete hearing aid for a more substantial hearing loss. A review is given on the current work on acoustically "transparent" open-fitting earpieces, algorithms for interaural cue enhancement, and for automatically detecting relevant changes in the acoustical scene that call for an altered processing strategy. The current prototype runs on the binaural, cable-connected master hearing aid (MHA) that includes earpieces that allow for approaching acoustic transparency. This can be achieved by individual calibration of the earpieces using in-situ in-ear measurements and electro-acoustic models of the earpieces and the ear canal. A binaural high-fidelity enhancement algorithm motivated by interaural magnification is evaluated in its benefit for normal and near-to-normal listeners. Finally, several acoustic event detection schemes are compared with respect to their applicability in real-life situations. They decompose the ambient acoustic signal into spectro-temporal basis functions and utilize machine learning classifiers to detect typical natural and artificial events. [Work supported by DFG.]

10:30

5aSP8. The effects of head-worn attire on measured head-related transfer functions. Stephen Nimick (Beats Electronics, LLC, 3201 Sawtelle Blvd., Apt. 108, Los Angeles, CA 90066, stephen.nimick@beatsbydre.com), Joshua Atkins, and Ismael Nawfal (Beats Electronics, LLC, Santa Monica, CA)

In this paper, we investigate measured changes in spatial perception presented by hats and other head-worn equipment. Particularly, helmets worn in sports, military and civil services (firefighters, police etc.) can affect spatial awareness and localization of wearers, possibly putting the wearer and others in danger. For this paper, a variety of hats were placed on a Head And Torso Simulator (HATS). The Head Related Transfer Function (HRTF) was measured in an anechoic chamber at a variety of angles around the HATS. The data will be analyzed to show the differences between various head-worn devices.

10:45

5aSP9. The effects of spectral fine structure and amplitude envelope on perceived similarity and identification of approaching vehicles. Jeremy Gaston (Army Res. Lab., Aberdeen Proving Grounds, Aberdeen, MD, jeremy.r.gaston.civ@mail.mil), Kelly Dickerson (Army Res. Lab., Abingdon, MD), and Mark Ericson (Army Res. Lab, Aberdeen, MD)

The recognition and identification of important complex natural sound sources is a critical component of maintaining environmental awareness. The properties of sound sources are not static; environmental factors can cause dramatic shifts in acoustic structure. The present study investigates the effects of large imposed acoustic changes on listener perception of approaching ground and aerial vehicles. In the first experiment, listeners rated the perceived similarity of the vehicle stimuli that were either (1) unaltered, or had (2) spectral fine structure replaced with random noise, or (3) had their natural amplitude envelopes flattened. The results of a multidimensional scaling (MDS) analysis found that the ordering of perceptual space for each of the conditions was generally a function of the acoustic information available. The second experiment investigated listener identification for the same sets of sounds. Identification performance across vehicle sounds had good agreement with similarity mappings and was best when spectral fine structure was smoothed and variations in amplitude envelope were maintained, and worst when the opposite was true. These results are consistent with the notion that amplitude envelope is a critical component in the recognition and identification of complex natural sound sources.

11:00

5aSP10. Assessment of heart rate variability from speech analysis. Nivedita Deshpande (School of Studies in Electronics & Photonics, Pt. Ravishankar Shukla Univ., Raipur, India), Kavita Thakur (School of Studies in Electronics & Photonics, Pt. Ravishankar Shukla Univ.,Raipur, Chhattisgarh 492010, India, kavithakur@rediffmail.com), and Arun S. Zadgaonkar (Electronics and TeleCommun. Eng., Dr C.V.Raman Univ., Bilaspur, Chhattisgarh, India)

In this paper, various methods of heart beat variability assessment from speech analysis have been presented. Heart rate variability (HRV) is a physiological phenomenon where the time interval between heart beats varies. It is measured by the variation in the beat-to-beat interval. The present work deals with the HRV detection from speech parameters. RR-cycle detects one heart beat. Continuous monitoring of electrocardiograph for a span of time can provide HRV. More than 250 samples of normal informants as well as heart patients with enlarged heart have been collected during the four years of research work. The regular ECG and speech samples of the patient have been collected and analyzed. Further they had compared with the parameters of a normal healthy informant. Speech samples were collected through a microphone and subjected to be digitized. The required speech segmental have been extracted and analyzed through a DSP tool, PRAAT. ECG sample has been recorded through an ECG machine. A technique of HRV detection from speech analysis has been presented in this paper. The HRV detection from speech can be a very helpful tool in monitoring the functioning of human heart.

11:15

5aSP11. Temporal stability of long-term measures of fundamental frequency. Pablo Arantes (Universidade Federal de São Carlos, Via Washington Luís, Km. 235 - Caixa Postal 676, São Carlos 13565-905, Brazil, pabloarantes@gmail.com) and Anders Eriksson (Dept. of Philosophy, Linguist and Theory of Sci., Univ. of Gothenburg, Gothenburg, Sweden)

We investigated long-term mean, median, and base value of the voice fundamental frequency (F0) to estimate how long it takes their variability to stabilize. That information can be useful in the development of F0 contour normalization and forensic applications. Change point analysis was used to locate changes in underlying variance in the mean, median, and base value time-series. In one experiment, stabilization points were calculated in recordings of the same text spoken in 26 languages. Average stabilization points are 5 s for base value and 10 s for mean and median. Variance after the stabilization point was reduced around 40 times for mean and median and more than 100 times for the base value. In another experiment, four speakers read two different texts each. Stabilization points for the same speaker across the texts do not exactly coincide as would be ideally expected. Average point dislocation is 2.5 s for the base value, 3.4 for the median, and 9.5 for the mean. After stabilization, differences in the three measures obtained from the two texts are 2% on average across speakers. Present results show that stabilization points in long-term measures of F0 occur earlier than suggested in the previous literature.

11:30

5aSP12. Improvement of speech coding algorithm based on DSP system. Xiaochen Wu (Harbin Eng. Univ., 803, Shushing Bldg., Harbin 150000, China, wuxiaochen629@163.com) and HaiNing Lv (Harbin Eng. Univ., Harbin, Heilongjiang Province, China)

With the rapid development of the communication technology, the high quality of speech communication has become one of the main development trends. In order to solve the quality problem in the processing of communication, pre emphasis is used to improve the quality of the high frequency components in codec algorithm. The speech enhancement of spectral subtraction is also applied in this speech coding algorithm to improve the ability of anti noise. Repackaging the sending frames and redefining the length of the frame that need to be send in the algorithm in order to realize the low speed in speech communication process. At last, the speech quality will be proved by PESQ—MOS test and the spectrogram. From the test and the figure, the quality of the synthesized speech especially with background noise has been significantly improved.

11:45

5aSP13. Acoustic transfer functions and speech intelligibility in permanent wearing headsets. Vladimir G. Popov (Lab. of Phys. of Semiconductor Nanostructures, Inst. of Microelectronics Technol. of Russian Acad. of Sci., Academician Osipyan St. 6, Chernogolovka 142432, Russian Federation, popov@iptm.ru), Alexey L. Ushakov (Necktec, Moscow, Russian Federation), Zintars Laris, Serge Batov, and Yurii Saprovskii (R&D Akustika, Riga, Latvia)

A detailed analysis of the acoustic transfer functions has been investigated for different locations of the headset microphones in the area of human head. Amplitude-frequency and phase-frequency characteristics of acoustic signals are discussed in the report. The signals have been recorded with the micro-mechanical-electrical-system (MEMS) microphones located on a mannequin, simulating the acoustic properties of the human torso. The the source of the signals is an emitter built in the mouth of the manikin's head. Analysis of the characteristics has shown that the acoustic signals undergo significant phase and amplitude changes associated with the interference and diffraction processes. These changes result in significant nonlinearity phase characteristics, and this nonlinearity depends on the installation location of the MEMS microphone. Also quality of speech intelligibility has been estimated by the method of expert evaluation and the speech has been recorded for different positions of the microphones and speaker head. A comparative analysis of the results has shown the optimal location of the microphones in the ear and on the breast of man.

FRIDAY MORNING, 9 MAY 2014 556 A/B, 8:00 A.M. TO 12:00 NOON

Session 5aUW

Underwater Acoustics: Underwater Acoustic Propagation

Ilya A. Udovydchenkov, Chair
Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., MS #9, Woods Hole, MA 02543

Contributed Papers

8:00

5aUW1. Modeling three-dimensional propagation in a continental shelf environment using the finite element method. Benjamin Goldsberry and Marcia J. Isakson (Appl. Res. Labs., The Univ. of Texas at Austin, 10000 Burnet Rd., Austin, TX 78758, bmgoldsberry@gmail.com)

Horizontal refraction is a well known three-dimensional propagation effect in underwater continental shelf environments. However, only a few experimental observations of this phenomenon exist. In 2007, horizontal refraction was not only experimentally verified off the coast of Florida but was found to dominate the acoustic propagation. Numerical models that allow for azimuthal variations in the geometry are needed to accurately model this environment. In this work, a three-dimensional finite element method with the longitudinally invariant technique is presented. In this method, a cosine transform is applied to the three-dimensional Helmholtz equation to remove the range-independent dimension. The finite element method is used to solve the transformed Helmholtz equation for each out-of-plane wavenumber. The inverse cosine integral is then used to transform the pressure field back to three-dimensional spatial coordinates. The computed pressure field is then suitable for comparison to experimental measurements. [Work supported by ONR, Ocean Acoustics.]

8:15

5aUW2. Explanation of an extended arrival structure measured in the Catoche Tongue using a three-dimensional propagation model. Megan S. Ballard, Jason D. Sagers, and David P. Knobles (Appl. Res. Labs. at the Univ. of Texas at Austin, P.O. Box 8029, Austin, TX 78758, meganb@arlut.utexas.edu)

This paper presents data recorded on a sparsely populated vertical line array (VLA) moored in the center of the Catoche Tongue, a major reentrant in the Campeche Bank in the southeastern Gulf of Mexico. An extended arrival structure observed in the pressure time series data resulting from signals underwater sound (SUS) deployed 50 to 80 km from the VLA is examined. Two distinct classes of arrivals are associated with the reception of each SUS event. The first class of arrivals was identified as resulting from the direct path, and the second class of arrivals is believed to result from horizontal refraction from the margins of the tongue. The difference in time between the onset of the first and second class of arrivals decreases as the distance from the VLA increases in a manner that is consistent with a decrease in the length of the refracted path relative to the direct path as suggested by the environment geometry. The observations are further supported by a three-dimensional (3D) acoustic propagation computation which reproduces many of the features of the measured data and provides additional insight into the details of the 3D propagation. [Work supported by ONR.]

8:30

5aUW3. Numerical modeling of three-dimensional underwater sound propagation under rough sea surfaces. Ying-Tsong Lin (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Bigelow 213, MS#11, WHOI, Woods Hole, MA 02543, ytlin@whoi.edu)

A three-dimensional parabolic-equation (PE) solution of long-distance underwater sound propagation with the influence of arbitrary sea surface roughness is derived. A stair-step approximation is implemented to the sea surface height along the PE solution marching direction. A higher-order operator splitting algorithm is utilized for an Alternative Direction Implicit (ADI) method to achieve computational efficiency and accuracy. The ADI method also ensures that the pressure release interface condition on the irregular sea surface is satisfied exactly. An theoretical solution will be used to benchmark the PE solution, and examples of sound propagation under sea surface wave swells will be shown. Depending on the source position relative to the waves, the acoustic waveguide condition can change from ducting to anti-ducting. Two-way PE solutions to include acoustic backscattering from rough sea surfaces will also be discussed. [Work supported by the ONR.]

8:45

5aUW4. Horizontal refraction and whispering gallery sound waves in area of curvilinear coastal wedge in shallow water. Boris Katsnelson (Marine GeoSci., Univ. of Haifa, Mt Carmel, Haifa 31905, Israel, katz@phys.vsu.ru) and Andrey Malykhin (Phys. Dept, Voronezh Univ., Voronezh, Russian Federation)

Horizontal refraction of sound signals in area of coastal wedge determines specific structure of the sound field in horizontal plane, including multipath areas and caustics (in the framework of approximation of horizontal rays and vertical modes). This structure depends on mode number and frequency. In the paper sound field of signal, radiated by the point source is studied in the shallow water area which can be considered as a wedge with curvilinear coastal line (lake, gulf, cape, etc.). It is shown that for some distances from the source to the beach whispering gallery waves (whispering gallery horizontal rays) exist, propagating along the coast, where energy is concentrated in narrow area in horizontal plane. Structure and characteristic of these rays depend on mode number, frequency and waveguide parameters (in particular, steepness of sea floor and radius of curvature). Intensity fluctuations as well as variation of pulse shape and frequency spectrum are studied; analytical estimations are presented as well as discussion of possible experimental observations. [Work was supported by RFBR and BSF.]

9:00

5aUW5. The effect of large- and small-scale sound velocity field structure on shallow water sound propagation in the East China Sea. Clare G. Nadig (Graduate Program in Acoust., The Penn. State Univ., 201 Appl. Sci. Bldg., University Park, PA 16802, cgn5004@psu.edu) and David L. Bradley (Appl. Res. Lab., The Penn. State Univ., State College, PA)

During the Transverse Acoustic Variability Experiment (TAVEX) in 2008, towed CTD measurements were used to record the sound speed field over an area of the East China Sea surrounding 34- and 20-km acoustic paths. The data revealed several propagating internal wave packets along with sound speed variability due to density-compensated thermohaline variations (spice). Concurrent acoustic data were also recorded from 300- and 500-Hz sources. Parabolic equation modeling is used to examine the effects of the measured internal waves and spice on acoustic propagation and to make comparisons with the recorded acoustic data.

9:15

5aUW6. Relationships between acoustic surface wave shape inversion and receiver array statistics in a variety of sea states. Sean Walstead and Grant Deane (ECE/SIO, UCSD, 9500 Gilman Dr., 0407, La Jolla, CA 92093-0407, swalstead@ucsd.edu)

Mid-frequency (12 kHz) acoustic scattering from the sea surface is analyzed with regard to surface wave shape inversion and receiver array characteristics. The forward scattered data is from the Surface Processes and Communications Experiment (SPACE08). Multipath arrivals representing surface, bottom-surface, and surface-bottom paths are distinguishable, implying that knowledge of the surface is known approximately 1/3, 1/2, and 2/3 the distance between source and receiver. Surface wave shape inversion at those ranges are presented in a variety of environmental conditions and compared to other calculable statistics such as vertical and horizontal array correlation length. The relationship between temporal fluctuations in the intensity of the surface reflected multi-path and array spatial coherence is investigated. To what extent receiver array statistics related to acoustic focusing follows environmental conditions such as wind speed and sea state are considered.

9:30

5aUW7. Sound propagation in a model ocean with internal waves: experiments and simulations. Likun Zhang, Santiago J. Benavides, and Harry L. Swinney (Dept. of Phys. and Ctr. for Nonlinear Dynam., Univ. of Texas at Austin, 2515 Speedway, Stop C1610, Austin, TX 78712-1199, lzhang@chaos.utexas.edu)

We conduct laboratory experiments and numerical simulations in a laboratory tank where sound waves and internal gravity waves are simultaneously generated and detected. The tank contains a density-stratified fluid with internal gravity waves produced by a wave generator. Acoustic sources and receivers are arranged such that the acoustic track crosses the internal wave beam. The internal wave energy density, energy flux, and total power are determined from particle imaging velocimetry measurements. The internal wave field is also computed in direct numerical simulations of the Navier-Stokes equations, using a parallelized finite-volume based solver code. The fluctuations in sound speed and intensity are determined as a function of the acoustic track location and the internal wave amplitude, frequency, phase, and modulation frequency. This research is designed to achieve a better understanding of sound propagation and scattering by internal waves in the oceans. [This research was supported by ONR MURI Grant N000141110701 (WHOI). L.Z. acknowledges the support of the 2013-14 ASA F. V. Hunt Postdoctoral Research Fellowship.]

9:45

5aUW8. Amplitude and phase fluctuations of the sound fled in vicinity of horizontally stratified perturbation in shallow water (parabolic equation modeling). Jixing Qin (State Key Lab. of Acoust., Inst. of Acoust., Chinese Acad. of Sci., Beijing, China) and Boris Katsnelson (Marine Geo-Sci., Univ. of Haifa, 1, Universitetskaya sq, Voronezh 394006, Russian Federation, katz@phys.vsu.ru)

Based on the approach of vertical modes and the parabolic equation (PE) in horizontal plane, three-dimensional (3D) effects in propagation of sound signals are considered: structure variation of broadband signal in the presence of horizontal stratification, such as coastal wedge area, temperature front, and nonlinear internal waves. Main factor, influencing on sound propagation is dependence of horizontal refraction on frequency and mode number. The following effect are studied: interference between direct and reflected signals in multipath area of horizontal plane for different frequencies and mode numbers, different directions of amplitude and phase front, evolution of spectrum and shape of pulse. [Work was supported by RFBR and BSF.]

10:00–10:15 Break

10:15

5aUW9. An investigation into the bottom interface treatment in parabolic equation models utilizing split-step Fourier and finite-difference algorithms. Kevin B. Smith (Dept. of Phys., Naval Postgrad. School, 833 Dyer Rd., Bldg. 232, Rm. 114, Monterey, CA 93943, kbsmith@nps.edu), David J. Thomson (733 Lomax Rd., Victoria, BC, Canada), and Paul Hursky (Heat, Light and Sound Res., Inc., San Diego, CA)

Recent analysis of parabolic equation model results computed using split-step Fourier algorithms have shown increasing phase errors at long range in simple, isospeed waveguides. Preliminary results suggest that these errors are due to the treatment of the density discontinuity at the water/bottom interface through the use of smoothing functions. In this work, several different approaches are investigated for improving the phase accuracy of the solution, including hybrid methods employing split-step Fourier and finite-difference algorithms, as well as equivalent bottom characterizations. The results are compared relative to their improvements in accuracy as well as computational efficiency.

10:30

5aUW10. Bottom interacting acoustics in the north Pacific. Ralph A. Stephen (Woods Hole Oceanogr. Inst., 360 Woods Hole Rd., Woods Hole, MA 02543-1592, rstephen@whoi.edu), Peter F. Worcester (Scripps Inst. of Oceanogr., La Jolla, CA), Ilya A. Udovydchenkov (Woods Hole Oceanographic Inst., Woods Hole, MA), and Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., La Jolla, CA)

During the 2004 Long-range Ocean Acoustic Propagation Experiment (LOAPEX), a new class of acoustic arrivals was observed on ocean bottom seismometers (OBSs) for ranges from 500 to 3200 km. The arrivals were called deep seafloor arrivals (DSFAs), because they were the dominant arrivals on the ocean bottom seismometers (OBSs), but were very weak on the deep vertical line array (Deep VLA), located above 750 m from the seafloor. Stephen *et al*. [JASA **134**, 3307–3317 (2013)] attributed some of these arrivals to bottom-diffracted, surface-reflected (BDSR) energy that scattered from a seamount near the Deep VLA and subsequently reflected from the sea surface before arriving at the OBSs. In the Ocean Bottom Seismometer Augmentation in the North Pacific (OBSANP) Experiment in June to July 2013, we returned to the Deep VLA site with a near-seafloor Distributed Vertical Line Array (DVLA) that extended upward 1000 m from the seafloor and 12 OBSs. We transmitted to the instruments with a ship-suspended J15-3 acoustic source. The receiver locations and transmission program were designed to test the hypothesis that DSFAs correspond to BDSR energy, to further define the characteristics of the DSFAs, and to understand the conditions under which DSFAs are excited and propagate.

10:45

5aUW11. Weakly dispersive modal pulse propagation in the North Pacific Ocean. Ilya A. Udovydchenkov (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., MS #9, Woods Hole, MA 02543, ilya@whoi.edu), Michael G. Brown (Appl. Marine Phys., Rosenstiel School of Marine and Atmospheric Sci., Miami, FL), Timothy F. Duda (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., Woods Hole, MA), Peter F. Worcester, Matthew A. Dzieciuch (Scripps Inst. of Oceanogr., Univ. of California at San Diego, La Jolla, CA), James A. Mercer, Rex K. Andrew (Appl. Phys. Lab., Univ. of Washington, Seattle, WA), Bruce M. Howe (Dept. of Ocean and Resources Eng., Univ. of Hawai, Honolulu, HI), and John A. Colosi (Dept. of Oceanogaphy, Naval Postgrad. School, Monterey, CA)

The propagation of weakly dispersive modal pulses is investigated using data collected during the 2004 long-range ocean acoustic propagation experiment (LOAPEX). Weakly dispersive modal pulses are characterized by weak dispersion- and scattering-induced pulse broadening; such modal pulses experience minimal propagation-induced distortion and are thus well suited to communications applications. In the LOAPEX environment, modes 1, 2, and 3 are approximately weakly dispersive. Using LOAPEX observations it is shown that, by extracting the energy carried by a weakly dispersive modal pulse, a transmitted communications signal can be recovered without performing channel equalization at ranges as long as 500 km; at that range a majority of mode 1 receptions have bit error rates (BERs) less than 10%, and 6.5% of mode 1 receptions have no errors. BERs are estimated for low order modes and compared with measurements of signal-to-noise ratio (SNR) and modal pulse spread. Generally, it is observed that larger modal pulse spread and lower SNR result in larger BERs. [Work supported by ONR.]

11:00

5aUW12. Weakly dispersive modal pulses in long-range underwater acoustic communications. Ilya A. Udovydchenkov (Appl. Ocean Phys. and Eng., Woods Hole Oceanogr. Inst., 266 Woods Hole Rd., MS #9, Woods Hole, MA 02543, ilya@whoi.edu)

Weakly dispersive modal pulses are contributions to acoustic wave field from individual modes that are characterized by weak dispersion- and scattering-induced broadening. These pulses experience little propagation-induced distortions and are suitable for communications applications. This work investigates, using numerical simulations, how to design a communication system that takes advantage of the physics of weakly dispersive modal pulses. Two groups of weakly dispersive modal pulses are identified in typical mid-latitude ocean environments: the lowest order modes, and mode numbers with the waveguide invariant being near-zero (often around mode 20 at 75 Hz). This work analyzes the source and receiving array requirements for achieving low bit error rates (BERs) in a binary communication without performing channel equalization. It is shown that low BERs are achieved with only 3 hydrophones for mode 1 processing at 500 km and with 30 hydrophones for mode 20 at 400 km range with good signal-to-noise ratio (SNR). It is demonstrated that if depths of hydrophones are allowed to vary with the source-receiver distance, 2 hydrophones are often sufficient to achieve low BERs even with intermediate mode numbers. Thus, full modal resolution is often unnecessary to achieve low BERs. The effects of variable SNR are also studied. [Work supported by ONR.]

11:15

5aUW13. Leveraging spatial diversity to mitigate interference in underwater acoustic communication networks. James McGee (Code 15 Sensors and SONAR Systems, Naval Undersea Warfare Ctr., 1176 Howell St., Newport, RI 02841, james.a.mcgee@navy.mil), Peter Swaszek (Elec., Comput. and Biomedical Eng., Univ. of Rhode Island, Kingston, RI), and Josko Catipovic (Code 15 Sensors and SONAR Systems, Naval Undersea Warfare Ctr., Newport, RI)

Many acoustic communication channels suffer from interference which is neither narrowband nor impulsive. This relatively long duration partial band interference can be particularly detrimental to system performance. To address the problem of underwater communications over a time-varying multipath channel in the presence of partial band interference, we examine a receiver which leverages the spatial diversity implicit in a network of geographically distributed hydrophones due to the slow speed of sound propagation. The network consists of multiple cabled hydrophones which receive communication signals from multiple users in addition to interfering signals from active sonars and marine mammals. The partial band interference corrupts different portions of the received signal depending on the relative position of the interferers, information source and receivers. The need for explicit time alignment and channel compensation due to the differing propagation paths between sources and receivers complicates combining the received signals. After surveying recent work in interference mitigation and orthogonal division multiplexing as background motivation, we examine the problem of combining information from such differentially corrupted signals through simulation.

11:30

5aUW14. Low frequency acoustic communication and the waveguide modal behavior during Shallow Water 2006 experiment. Mohsen Badiey and Aijun Song (College of Earth, Ocean, and Environment, Univ. of Delaware, 261 S. College Ave., Robinson Hall, Newark, DE 19716, badiey@udel.edu)

Low frequency (i.e., $<$500 Hz) acoustic intensity fluctuations in the presence of internal waves has been studied in some detail during the past two decades. However, not much has been reported on the relationship between the modal behavior of the waveguide and the potential for long-range communications. Previously, we have shown significant performance degradation at higher frequencies (i.e., 800 and 1600 Hz) while an internal wave packet travels through a source-receiver track. Here, we further examine the acoustic transmissions at lower frequencies (i.e., 100 and 200 Hz) during the same internal wave event. Significant intensity fluctuations at these frequencies can be explained by modal analysis. We also report acoustic communication performance at multiple frequencies (i.e., 100, 200, 800, and 1600 Hz). The frequency dependency is analyzed with the focus on modal behaviors to explain the performance variation of acoustic communication during the passage of the internal waves.

11:45

5aUW15. Multiple input multiple output underwater communication based on differential amplitude phase shift keying modulation. Xu Xia and Jingwei Yin (College of Underwater Acoust. Eng., Harbin Eng. Univ., Rm. 801, Acoust. Bldg., No.145 Nantong St., Harbin, Heilongjiang 150001, China, heaven663@163.com)

The MIMO-OFDM combining time, spatial, and frequency diversity can effectively improve channel capacity and transmission efficiency of underwater acoustic (UWA) communication system. Space-time coding needs a large number of pilot signals to estimate UWA channel at the receiving end, which increases the complexity of the systems and limits the communication rate. For this, space-time coding combined with differential amplitude phase shift keying modulation (DAPSK) is proposed in this paper. It can complete the decoding without any prior knowledge of UWA channel, reducing the complexity of the system, saving channel resources and improving the transmission efficiency. Simulation analysis on UWA MIMO-OFDM systems shows this algorithm is feasible, which provides a feasible method for high-speed transmission in UWA communication.

This document is frequently updated; the current version can be found online at the Internet site: <http://scitation.aip.org/content/asa/journal/jasa/info/authors>.

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, 2 Huntington Quadrangle, Suite 1NO1, Melville, NY 11747-4502

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

a)E-mail: jasa@aip.org

features that are desired for the submitted manuscript. This document extracts many of the style suggestions found in the *AIP Style Manual,*[1] which is available online at the internet site <http://www.aip.org/pubservs/style/4thed/toc.html>. The *AIP Style Manual,* although now somewhat dated and not specifically directed toward publication in the *Journal of the Acoustical Society of America* (JASA), is a substantially more comprehensive document, and authors must make use of it also when preparing manuscripts. If conflicting instructions are found in the two documents, those given here take precedence. (Authors should also look at recent issues of the *Journal* for examples of how specific style issues are handled.) Conscientious consideration of the instructions and advice given in the two documents should considerably increase the likelihood that a submitted manuscript will be rapidly processed and accepted for publication.

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.

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(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.

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(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, WordPerfect, or PS 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, or EPS (see also, Section V. H); one file for each cited figure number. PDF figures may be submitted but these are not desirable. (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.]

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(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, or TIFF format, then they will probably be used directly, at least in part. If they are submitted in PDF format, then they possibly may have to be printed out and then scanned to place them in the desired format for production editing.

D. Steps in online submission

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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.

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A. Mandatory charges

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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:[2]

[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,[3] 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. The *AIP Style Manual*[1] gives a lengthy list of standard abbreviations that are used for journals that report physics research, but the interdisciplinary nature of acoustics is such that the list omits many journals that are routinely cited in the *Journal of the Acoustical Society of America.* 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, 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*[4] (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[5] 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[6] 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*[7]) 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,[8] Merriam-Webster's Collegiate Dictionary, 11th Edition,[9] Strunk and White s Elements of Style,[10] and the Chicago Manual of Style.[11] (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,[12] or in a dictionary specifically devoted to acoustics such as the Dictionary of Acoustics[13] 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.

REFERENCES

[1] AIP Publication Board (R. T. Beyer, chair), *AIP Style Manual* (American Institute of Physics, 2 Huntington Quadrangle, Suite 1NO1, Melville, NY 11747, 1990, 4th ed.). This is available online at <http://www.aip.org/pubservs/style/4thed/toc.html>.

[2] M. Mellody and G. H. Wakefield, "The time-frequency characteristics of violin vibrato: Modal distribution analysis and synthesis," J. Acoust. Soc. Am. **107,** 598-611 (2000).

[3] See, for example, the paper: B. Møhl, M. Wahlberg, P. Madsen, L. A. Mller, and A. Surlykke, "Sperm whale clicks: Directionality and source level revisited," J. Acoust. Soc. Am. **107,** 638–648 (2000).

[4] American Institute of Physics, *Physics and Astronomy Classification Scheme 2003.* A paper copy is available from AIP Publishing LLC, 2 Huntington Quadrangle, Suite 1NO1, Melville, NY 11747. It is also available online at the site <http://www.aip.org/pacs/index.html>.

[5] A. D. Pierce, Current criteria for selection of articles for publication, J. Acoust. Soc. Am. **106,** 1613–1616 (1999).

[6] A. D. Pierce, Literate writing and collegial citing, J. Acoust. Soc. Am. **107,** 2303–2311 (2000).

[7] *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>.

[8] *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).

[9] *Merriam-Webster's Collegiate Dictionary, 11th Edition* (Merriam-Webster, Springfield, MA, 2003, principal copyright 1993). (A freshly updated version is issued annually.)

[10] W. Strunk, Jr. and E. B. White, *The Elements of Style,* with forward by Roger Angell (Allyn and Bacon, 1999, 4th edition).

[11] *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).

[12] *Academic Press Dictionary of Science and Technology,* edited by Christopher Morris (Academic Press, Inc., 1992).

[13] C. L. Morfey, *Dictionary of Acoustics* (Academic Press, Inc., 2000).

ASSOCIATE EDITORS IDENTIFIED WITH PACS CLASSIFICATION ITEMS

The Classification listed here is based on the Appendix to Section 43, “Acoustics,” of the current edition of the Physics and Astronomy Classification Scheme PACS of AIP Publishing LLC. The full and most current listing of PACS can be found at the internet site <http://www.aip.org/pubservs/pacs.html>. In the full PACS listing, all of the acoustics items are preceded by the primary classification number 43. The listing here omits the prefatory 43; a listing in the AIP Publishing document such as 43.10.Ce will appear here as 10.Ce.

The present version of the Classification scheme is intended as a guide to authors of manuscripts submitted to the Journal who are asked at the time of submission to suggest an Associate Editor who might handle the processing of their manuscript. Authors should note that they can also have their manuscripts processed from any of the special standpoints of (i) Applied Acoustics, (ii) Computational Acoustics, (iii) Mathematical Acoustics, or (iv) Education in Acoustics, and that there are special Associate Editors who have the responsibility for processing manuscripts from each of these standpoints.

The initials which appear in brackets following most of the listings correspond to the names of persons on the Editorial Board i.e., Associate Editors who customarily edit material that falls within that classification. A listing of full names and institutional affiliations of members of the Editorial Board can be found on the back cover of each issue of the *Journal*. A more detailed listing can be found at the internet site <http://asadl.org/jasa/for_authors_jasa>. The most current version of the present document can also be found at that site.

[05] Acoustical Society of America

05.Bp Constitution and bylaws [EM]
05.Dr History [ADP]
05.Ft Honorary members [EM]
05.Gv Publications ARLO. Echoes. ASA Web page, electronic archives and references [ADP]
05.Hw Meetings [EM]
05.Ky Members and membership lists, personal notes, fellows [EM]
05.Ma Administrative committee activities [EM]
05.Nb Technical committee activities; Technical Council [EM]
05.Pc Prizes, medals, and other awards [EM]
05.Re Regional chapters [EM]
05.Sf Obituaries

[10] General

10.Ce Conferences, lectures, and announcements (not of the Acoustical Society of America) [EM]
10.Df Other acoustical societies and their publications]; online journals and other electronic publications [ADP]
10.Eg Biographical, historical, and personal notes (not of the Acoustical Society of America) [EM]
10.Gi Editorials, Forum [ADP], [NX]
10.Hj Books and book reviews [PLM]
10.Jk Bibliographies [EM], [ADP]
10.Km Patents [DLR], [SAF]
10.Ln Surveys and tutorial papers relating to acoustics research, tutorial papers on applied acoustics [ADP], [NX]
10.Mq Tutorial papers of historical and philosophical nature [ADP], [NX], [WA]
10.Nq News with relevance to acoustics nonacoustical theories of interest to acoustics [EM], [ADP]
10.Pr Information technology, internet, nonacoustical devices of interest to acoustics [ADP], [NX]
10.Qs Notes relating to acoustics as a profession [ADP], [NX]
10.Sv Education in acoustics, tutorial papers of interest to acoustics educators [LLT], [WA], [BEA], [VWS], [PSW]
10.Vx Errata [ADP]

[15] Standards [SB], [PDS]

[20] **General linear acoustics**

20.Bi Mathematical theory of wave propagation [MD], [SFW], [ANN], [RM], [RKS], [KML], [CAS]
20.Dk Ray acoustics [JES], [SFW], [ANN], [JAC], [KML], [TFD], [TK]
20.El Reflection, refraction, diffraction of acoustic waves [JES], [OU], [SFW], [RM], [KML], [GH], [TFD], [TK]
20.Fn Scattering of acoustic waves [LLT], [JES], [OU], [SFW], [RM], [RMW], [KML], [RR], [GH], [TK]
20.Gp Reflection, refraction, diffraction, interference, and scattering of elastic and poroelastic waves [OU], [RM], [DF], [RKS], [JAT], [DSB], [GH]
20.Hq Velocity and attenuation of acoustic waves [MD], [OU], [SFW], [TRH], [RAS], [NPC], [JAT], [GH]
20.Jr Velocity and attenuation of elastic and poroelastic waves [ANN], [NPC], [RKS], [GH]
20.Ks Standing waves, resonance, normal modes [LLT], [SFW], [JHG], [JGM], [RM], [JDM]
20.Mv Waveguides, wave propagation in tubes and ducts [OU], [LH], [JHG], [RK], [JGM], [RMW], [JBL]
20.Px Transient radiation and scattering [LLT], [JES], [JHG], [JGM], [ANN], [RMW], [MDV], [DDE]
20.Rz Steady-state radiation from sources, impedance, radiation patterns, boundary element methods [SFW], [RM], [FCS], [JHG], [JGM]
20.Tb Interaction of vibrating structures with surrounding medium [LLT], [RM], [FCS], [JHG], [JGM], [LH]
20.Wd Analogies [JDM]
20.Ye Measurement methods and instrumentation [SFW], [TRH], [KGF], [JDM], [GH]

[25] Nonlinear acoustics

25.Ba Parameters of nonlinearity of the medium [MD], [OAS], [ROC]
25.Cb Macrosonic propagation, finite amplitude sound; shock waves [OU], [MDV], [PBB], [OAS], [ROC]
25.Dc Nonlinear acoustics of solids [MD], [ANN], [OAS]
25.Ed Effect of nonlinearity on velocity and attenuation [MD], [OAS], [ROC]
25.Fe Effect of nonlinearity on acoustic surface waves [MD], [MFH], [OAS]
25.Gf Standing waves; resonance [OAS], [MFH]
25.Hg Interaction of intense sound waves with noise [OAS], [PBB]
25.Jh Reflection, refraction, interference, scattering, and diffraction of intense sound waves [OU], [MDV], [PBB]
25.Lj Parametric arrays, interaction of sound with sound, virtual sources [TRH], [KGF], [RMW]
25.Nm Acoustic streaming [JDM], [OAS], [RMW], [LH]
25.Qp Radiation pressure [RMW], [ROC]
25.Rq Solitons, chaos [MFH]
25.Ts Nonlinear acoustical and dynamical systems [MFH], [ROC]
25.Uv Acoustic levitation [MFH], [KGF]
25.Vt Intense sound sources [RMW], [ROC], [TRH]
25.Yw Nonlinear acoustics of bubbly liquids [TGL], [SWY]
25.Zx Measurement methods and instrumentation for nonlinear acoustics [ROC]

[28] Aeroacoustics and atmospheric sound

28.Bj Mechanisms affecting sound propagation in air, sound speed in the air [DKW], [VEO], [KML], [RR]
28.Dm Infrasound and acoustic-gravity waves [RMW], [DKW], [PBB], [RR]
28.En Interaction of sound with ground surfaces, ground cover and topography, acoustic impedance of outdoor surfaces [OU], [KVH], [VEO], [KML], [RR]
28.Fp Outdoor sound propagation through a stationary atmosphere, meteorological factors [RMW], [DKW], [KML], [TK]
28.Gq Outdoor sound propagation and scattering in a turbulent atmosphere, and in non-uniform flow fields [VEO], [PBB], [KML]
28.Hr Outdoor sound sources [JWP], [PBB], [TK]
28.Js Numerical models for outdoor propagation [VEO], [NAG], [DKW]
28.Kt Aerothermoacoustics and combustion acoustics [AH], [JWP], [LH]
28.Lv Statistical characteristics of sound fields and propagation parameters [DKW], [VEO]
28.Mw Shock and blast waves, sonic boom [VWS], [ROC], [PBB], [RR]
28.Py Interaction of fluid motion and sound. Doppler effect and sound in flow ducts [JWP], [AH], [LH]
28.Ra Generation of sound by fluid flow, aerodynamic sound, and turbulence, [JWP], [AH], [PBB], [RR], [TK], [LH]
28.Tc Sound-in-air measurements, methods and instrumentation for location, navigation, altimetry, and sound ranging [JWP], [KVH], [DKW], [RR]
28.Vd Measurement methods and instrumentation to determine or evaluate atmospheric parameters, winds, turbulence, temperatures, and pollutants in air [JWP], [DKW], [RR]
28.We Measurement methods and instrumentation for remote sensing and for inverse problems [DKW]

[30] Underwater sound

30.Bp Normal mode propagation of sound in water [AMT], [MS], [NPC], [JIA], [TFD]
30.Cq Ray propagation of sound in water [JES], [JIA], [JAC], [TFD]
30.Dr Hybrid and asymptotic propagation theories, related experiments [JIA], [JAC], [TFD]
30.Es Velocity, attenuation, refraction, and diffraction in water, Doppler effect [DRD], [JAC], [JIA], [TFD]
30.Ft Volume scattering [KGF], [APL]
30.Gv Backscattering, echoes, and reverberation in water due to combinations of boundaries [JIA], [APL]
30.Hw Rough interface scattering [JES], [APL]
30.Jx Radiation from objects vibrating under water, acoustic and mechanical impedance [DSB], [DF], [EGW], [DDE]

30.Ky Structures and materials for absorbing sound in water; propagation in fluid-filled permeable material [NPC], [FCS], [TRH]

30.Lz Underwater applications of nonlinear acoustics; explosions [NAG], [OAS], [KGF], [SWY]

30.Ma Acoustics of sediments; ice covers, viscoelastic media; seismic underwater acoustics [NAG], [MS], [DSB], [JIA]

30.Nb Noise in water; generation mechanisms and characteristics of the field [KGS], [MS], [JAC], [SWY]

30.Pc Ocean parameter estimation by acoustical methods; remote sensing; imaging, inversion, acoustic tomography [KGS], [AMT], [MS], [JAC], [JIA], [ZHM], [HCS], [SED], [TFD], [APL]

30.Qd Global scale acoustics; ocean basin thermometry, transbasin acoustics [JAC]

30.Re Signal coherence or fluctuation to sound propagation/scattering in the ocean [KGS], [HCS], [TFD]

30.Sf Acoustical detection of marine life; passive and active [DKM], [AMT], [MS], [KGF], [MCH], [JIA], [APL]

30.Tg Navigational instruments using underwater sound [HCS], [JAC]

30.Vh Active sonar systems [JES], [TRH], [ZHM], [DDE]

30.Wi Passive sonar systems and algorithms, matched field processing in underwater acoustics [KGS], [HCS], [AMT], [MS], [SED], [JIA], [ZHM]

30.Xm Underwater measurement and calibration instrumentation and procedures [KGF], [JAC], [TRH], [DDE]

30.Yj Transducers and transducer arrays for underwater sound; transducer calibration [TRH], [DDE]

30.Zk Experimental modeling [JES], [MS], [TFD]

[35] Ultrasonics, quantum acoustics, and physical effects of sound

35.Ae Ultrasonic velocity, dispersion, scattering, diffraction, and attenuation in gases [RR], [GH], [TK]

35.Bf Ultrasonic velocity, dispersion, scattering, diffraction, and attenuation in liquids, liquid crystals, suspensions, and emulsions [DSB], [NAG], [JDM], [GH]

35.Cg Ultrasonic velocity, dispersion, scattering, diffraction, and attenuation in solids; elastic constants [MD], [MFH], [JDM], [JAT], [RKS], [GH], [TK]

35.Dh Pretersonics (sound of frequency above 10 GHz); Brillouin scattering [MFH], [RLW]

35.Ei Acoustic cavitation, vibration of gas bubbles in liquids [TGL], [NAG], [DLM]

35.Fj Ultrasonic relaxation processes in gases, liquids, and solids [RR], [NAG], [RMW]

35.Gk Phonons in crystal lattices, quantum acoustics [DF], [LPF], [JDM]

35.Hl Sonoluminescence [NAG], [TGL]

35.Kp Plasma acoustics [MFH], [JDM]

35.Lq Low-temperature acoustics, sound in liquid helium [JDM]

35.Mr Acoustics of viscoelastic materials [LLT], [MD], [OU], [FCS], [KVH], [GH]

35.Ns Acoustical properties of thin films [ADP], [TK]

35.Pt Surface waves in solids and liquids [MD], [ANN], [GH], [TK]

35.Rw Magnetoacoustic effect; oscillations and resonance [DAB], [DF], [LPF]

35.Sx Acoustooptical effects, optoacoustics, acoustical visualization, acoustical microscopy, and acoustical holography [JDM], [TK]

35.Ty Other physical effects of sound [MFH], [NAG]

35.Ud Thermoacoustics, high temperature acoustics, photoacoustic effect [RR], [JDM], [TB]

35.Vz Chemical effects of ultrasound [TGL]

35.Wa Biological effects of ultrasound, ultrasonic tomography [DLM], [MCH], [SWY]

35.Xd Nuclear acoustical resonance, acoustical magnetic resonance [JDM]

35.Yb Ultrasonic instrumentation and measurement techniques [ROC], [GH], [KAW], [TK]

35.Zc Use of ultrasonics in nondestructive testing, industrial processes, and industrial products [MD], [JAT], [ANN], [BEA], [GH], [TK]

[38] Transduction; acoustical devices for the generation and reproduction of sound

38.Ar Transducing principles, materials, and structures: general [MS], [DAB], [TRH], [DDE]

38.Bs Electrostatic transducers [MS], [KG], [DAB], [TRH], [MRB], [DDE]

38.Ct Magnetostrictive transducers [DAB], [TRH], [DDE]

38.Dv Electromagnetic and electrodynamic transducers [MS], [DAB], [TRH], [DDE]

38.Ew Feedback transducers [MS]

38.Fx Piezoelectric and ferroelectric transducers [DAB], [KG], [TRH], [MRB], [DDE]

38.Gy Semiconductor transducers [MS], [MRB]

38.Hz Transducer arrays, acoustic interaction effects in arrays [DAB], [TRH], [MS], [BEA], [MRB], [DDE]

38.Ja Loudspeakers and horns, practical sound sources [MS], [MRB], [DDE]

38.Kb Microphones and their calibration [MS], [MRB]

38.Lc Amplifiers, attenuators, and audio controls [MS]

38.Md Sound recording and reproducing systems, general concepts [MAH], [MRB]

38.Ne Mechanical, optical, and photographic recording and reproducing systems [MS]

38.Pf Hydroacoustic and hydraulic transducers [DAB]

38.Qg Magnetic and electrostatic recording and reproducing systems [MS]

38.Rh Surface acoustic wave transducers [MS], [TK]

38.Si Telephones, earphones, sound power telephones, and intercommunication systems [MS]

38.Tj Public address systems, sound-reinforcement systems [ADP]

38.Vk Stereophonic reproduction [ADP], [MRB]

38.Wl Broadcasting (radio and television) [ADP]

38.Yn Impulse transducers [MS]

38.Zp Acoustooptic and photoacoustic transducers [DAB], [MS]

[40] Structural acoustics and vibration

40.At Experimental and theoretical studies of vibrating systems [KML], [EGW], [DDE], [DF], [DAB], [FCS], [JHG], [JGM], [LC]

40.Cw Vibrations of strings, rods, and beams [DDE], [EGW], [DAB], [LPF], [JAT], [JHG], [JGM], [LC], [BEA]

40.Dx Vibrations of membranes and plates [LLT], [MD], [EGW], [DAB], [DF], [LPF], [JHG], [JGM], [LC], [JBL], [DDE]

40.Ey Vibrations of shells [DAB], [DF], [LPF], [JHG], [JGM], [EGW], [LC], [DDE]

40.Fz Acoustic scattering by elastic structures [LLT], [KML], [ANN], [DSB], [JHG], [TK], [JGM], [EGW], [DDE]

40.Ga Nonlinear vibration [AJMD], [RMW], [JAT], [JHG], [JGM]

40.Hb Random vibration TBA

40.Jc Shock and shock reduction and absorption [OU], [JHG], [JGM]

40.Kd Impact and impact reduction, mechanical transients [FCS], [JHG], [JGM]

40.Le Techniques for nondestructive evaluation and monitoring, acoustic emission [JAT], [BEA], [TK]

40.Ng Effects of vibration and shock on biological systems, including man [MCH]

40.Ph Seismology and geophysical prospecting; seismographs [MFH], [RKS], [ANN]

40.Qi Effect of sound on structures, fatigue; spatial statistics of structural vibration [JAT], [DDE]

40.Rj Radiation from vibrating structures into fluid media [LLT], [KML], [FCS], [EGW], [LC], [LH], [DDE]

40.Sk Inverse problems in structural acoustics and vibration [KML], [EGW], [LC], [DDE]

40.Tm Vibration isolators, attenuators, and dampers [LC]

40.Vn Active vibration control [BSC], [LC]

40.Yq Instrumentation and techniques for tests and measurement relating to shock and vibration, including vibration pickups, indicators, and generators, mechanical impedance [LC]

[50] Noise: its effects and control

50.Ba Noisiness: rating methods and criteria [GB], [SF], [BSF]

50.Cb Noise spectra, determination of sound power [GB], [KVH]

50.Ed Noise generation [KVH], [RK]

50.Fe Noise masking systems [BSF]

50.Gf Noise control at source: redesign, application of absorptive materials and reactive elements, mufflers, noise silencers, noise barriers, and attenuators, etc. [OU], [SFW], [RK], [FCS], [AH], [LC], [JBL], [LH]

50.Hg Noise control at the ear [FCS], [BSF]

50.Jh Noise in buildings and general machinery noise [RK], [KVH], [KML]

50.Ki Active noise control [BSC], [LC]

50.Lj Transportation noise sources: air, road, rail, and marine vehicles [GB], [SFW], [SF], [JWP], [KVH], [KML]

50.Nm Aerodynamic and jet noise [SF], [JWP], [AH], [LH]

50.Pn Impulse noise and noise due to impact [GB], [KVH], [SF]

50.Qp Effects of noise on man and society [GB], [BSF], [SF]

50.Rq Environmental noise, measurement, analysis, statistical characteristics [GB], [BSF], [SF]

50.Sr Community noise, noise zoning, by-laws, and legislation [GB], [BSF], [SF]

50.Vt Topographical and meteorological factors in noise propagation [PBB], [VEO]

50.Yw Instrumentation and techniques for noise measurement and analysis [GB], [KVH], [RK]

[55] Architectural acoustics

55.Br Room acoustics: theory and experiment; reverberation, normal modes, diffusion, transient and steady-state response [MV], [JES], [LMW], [FCS]

55.Cs Stationary response of rooms to noise; spatial statistics of room response; random testing [MV], [JES], [LMW]

55.Dt Sound absorption in enclosures: theory and measurement; use of absorption in offices, commercial and domestic spaces [MV], [JES], [LMW], [FCS]

55.Ev Sound absorption properties of materials: theory and measurement of sound absorption coefficients; acoustic impedance and admittance [MV], [OU], [FCS]

55.Fw Auditorium and enclosure design [MV], [JES], [LMW], [NX]

55.Gx Studies of existing auditoria and enclosures [MV], [JES], [LMW]

55.Hy Subjective effects in room acoustics, speech in rooms [MV], [JES], [MAH]

55.Jz Sound-reinforcement systems for rooms and enclosures [MV], [LMW], [MAH]

55.Ka Computer simulation of acoustics in enclosures, modeling [LLT], [MV], [JES], [SFW], [NAG]

55.Lb Electrical simulation of reverberation [MV], [LMW], [MAH]

55.Mc Room acoustics measuring instruments, computer measurement of room properties [MV], [JES], [LMW]

55.Nd Reverberation room design: theory, applications to measurements of sound absorption, transmission loss, sound power [MV], [LMW]

55.Pe Anechoic chamber design, wedges [ADP]

55.Rg Sound transmission through walls and through ducts: theory and measurement [LLT], [FCS], [LC], [BEA]

55.Ti Sound-isolating structures, values of transmission coefficients [LLT], [LC]

55.Vj Vibration-isolating supports in building acoustics [ADP]

55.Wk Damping of panels [LLT]

[58] Acoustical measurements and instrumentation

58.Bh Acoustic impedance measurement [DAB], [FCS]

58.Dj Sound velocity [RR], [JIA], [DKW], [TB], [GH], [TK]

58.Fm Sound level meters, level recorders, sound pressure, particle velocity, and sound intensity measurements, meters, and controllers [MS], [DKW], [TB], [KAW]

58.Gn Acoustic impulse analyzers and measurements [ADP]

58.Hp Tuning forks, frequency standards; frequency measuring and recording instruments; time standards and chronographs [MS]

58.Jq Wave and tone synthesizers [MAH]

58.Kr Spectrum and frequency analyzers and filters; acoustical and electrical oscillographs; photoacoustic spectrometers; acoustical delay lines and resonators [ADP]

58.Ls Acoustical lenses and microscopes [ADP]

58.Mt Phase meters [ADP]

58.Pw Rayleigh disks [ADP]

58.Ry Distortion: frequency, nonlinear, phase, and transient; measurement of distortion [MS]

58.Ta Computers and computer programs in acoustics [FCS], [DSB], [VWS]

58.Vb Calibration of acoustical devices and systems [DAB]

58.Wc Electrical and mechanical oscillators [ADP]

[60] Acoustic signal processing

60.Ac Theory of acoustic signal processing [KGS], [SAF], [MAH]

60.Bf Acoustic signal detection and classification, applications to control systems [JES], [SAF], [MRB], [PJL], [ZHM], [MAH], [JAC]

60.Cg Statistical properties of signals and noise [KGS], [MAH], [SAF], [TFD]

60.Dh Signal processing for communications: telephony and telemetry, sound pickup and reproduction, multimedia [MAH], [SAF], [HCS], [MRB]

60.Ek Acoustic signal coding, morphology, and transformation [MAH]

60.Fg Acoustic array systems and processing, beam-forming [JES], [ZHM], [HCS], [AMT], [MRB], [BEA], [TFD]

60.Gk Space-time signal processing other than matched field processing [JES], [ZHM], [JAC], [MRB]

60.Hj Time-frequency signal processing, wavelets [KGS], [SAF], [ZHM], [CAS], [PJL]

60.Jn Source localization and parameter estimation [JES], [KGS], [MAH], [ZHM], [MRB], [SED]

60.Kx Matched field processing [AIT], [AMT], [SED]

60.Lq Acoustic imaging, displays, pattern recognition, feature extraction [JES], [KGS], [SAF], [BEA], [MRB]

60.Mn Adaptive processing [SAF], [JIA], [DKW], [MRB]

60.Np Acoustic signal processing techniques for neural nets and learning systems [MAH], [AMT]

60.Pt Signal processing techniques for acoustic inverse problems [ZHM], [MRB], [SED]

60.Qv Signal processing instrumentation, integrated systems, smart transducers, devices and architectures, displays and interfaces for acoustic systems [MAH], [MRB]

60.Rw Remote sensing methods, acoustic tomography [DKW], [JAC], [ZHM], [AMT]

60.Sx Acoustic holography [JDM], [OAS], [EGW], [MRB]

60.Tj Wave front reconstruction, acoustic time-reversal, and phase conjugation [OAS], [HCS], [EGW], [BEA], [MRB]

60.Uv Model-based signal processing [ZHM], [MRB], [PJL]

60.Vx Acoustic sensing and acquisition [MS], [SAF], [JIA], [DKW]

60.Wy Non-stationary signal analysis, non-linear systems, and higher order statistics [EJS], [SAF], [PJL]

[64] Physiological acoustics

64.Bt Models and theories of the auditory system [ICB], [FCS], [CAS], [CA], [ELP]

64.Dw Anatomy of the cochlea and auditory nerve [AMS], [ANP], [SFW], [RRF], [CAS], [CA]

64.Fy Anatomy of the auditory central nervous system [AMS], [ANP], [RRF], [CAS], [CA]

64.Gz Biochemistry and pharmacology of the auditory system [CAS], [CA]

64.Ha Acoustical properties of the outer ear; middle-ear mechanics and reflex [FCS], [CAS], [CA], [ELP]

64.Jb Otoacoustic emissions [MAH], [CAS], [CA], [ELP]

64.Kc Cochlear mechanics [KG], [CAS], [CA], [ELP]

64.Ld Physiology of hair cells [KG], [CAS], [CA], [ELP]

64.Me Effects of electrical stimulation, cochlear implant [ICB], [CAS], [CA], [ELP]

64.Nf Cochlear electrophysiology [ICB], [KG], [CAS], [CA], [ELP]

64.Pg Electrophysiology of the auditory nerve [AMS], [ICB], [CAS], [CA], [ELP]

64.Qh Electrophysiology of the auditory central nervous system [AMS], [ICB], [CAS], [ELP]

64.Ri Evoked responses to sounds [ICB], [CAS], [CA], [ELP]

64.Sj Neural responses to speech [ICB], [CAS], [ELP]

64.Tk Physiology of sound generation and detection by animals [AMS], [MCH], [CAS]

64.Vm Physiology of the somatosensory system [MCH]

64.Wn Effects of noise and trauma on the auditory system [ICB], [CAS], [ELP]

64.Yp Instruments and methods [KG], [MAH], [CAS]

[66] Psychological acoustics

66.Ba Models and theories of auditory processes [EB], [CAS], [ELP], [JFC]

66.Cb Loudness, absolute threshold [MAS], [ELP]

66.Dc Masking [VMR], [EAS], [FJG], [LRB], [EB], [ELP], [JFC]

66.Ed Auditory fatigue, temporary threshold shift [EAS], [MAS], [ELP], [EB]

66.Fe Discrimination: intensity and frequency [VMR], [FJG], [EB]

66.Gf Detection and discrimination of sound by animals [ADP]

66.Hg Pitch [ADP]

66.Jh Timbre, timbre in musical acoustics [DD]

66.Ki Subjective tones [JFC]

66.Lj Perceptual effects of sound [VMR], [VB], [DB], [EB], [JFC]

66.Mk Temporal and sequential aspects of hearing; auditory grouping in relation to music [EAS], [FJG], [DB], [EB], [DD]

66.Nm Phase effects [EB], [JFC]

66.Pn Binaural hearing [VB], [LRB], [EB], [ELP], [NAG], [JFC]

66.Qp Localization of sound sources [VB], [FJG], [LRB], [EB], [ELP], [JFC]

66.Rq Dichotic listening [FJG], [LRB], [EB], [DD], [ELP], [JFC]

66.Sr Deafness, audiometry, aging effects [DS], [FJG], [ICB], [MAS], [ELP], [JFC]

66.Ts Auditory prostheses, hearing aids [DB], [VB], [FJG], [ICB], [MAS], [JFC], [EB], [ELP]

66.Vt Hearing protection [FCS]

66.Wv Vibration and tactile senses [MCH]

66.Yw Instruments and methods related to hearing and its measurement [ADP]

[70] Speech production

70.Aj Anatomy and physiology of the vocal tract, speech aerodynamics, auditory kinetics [BHS], [ZZ], [CYE], [CHS], [SSN], [LK]

70.Bk Models and theories of speech production [BHS], [ZZ], [CYE], [CHS]

70.Dn Disordered speech [BHS], [ZZ], [CYE], [LK] [CHS], [DAB]

70.Ep Development of speech production [CYE], [DAB], [CHS], [BRM], [BHS], [ZZ], [LK]

70.Fq Acoustical correlates of phonetic segments and suprasegmental properties: stress, timing, and intonation [CYE], [SSN], [DAB], [CGC], [SAF], [BHS]

70.Gr Larynx anatomy and function; voice production characteristics [CYE], [CHS], [LK], [SAF], [BHS], [ZZ]

70.Jt Instrumentation and methodology for speech production research [DAB], [CHS], [LK], [BHS], [ZZ]

70.Kv Cross-linguistics speech production and acoustics [DAB], [SAF], [BRM], [LK]

70.Mn Relations between speech production and perception [CYE], [DAB], [CHS], [CGC], [BRM], [BHS], [ZZ]

[71] Speech perception

71.An Models and theories of speech perception [ICB], [MAH], [CGC]

71.Bp Perception of voice and talker characteristics [CGC], [JHM], [BRM], [MSV], [MAH]

71.Es Vowel and consonant perception; perception of words, sentences, and fluent speech [DB], [PBN], [CGC], [BRM], [MAH]

71.Ft Development of speech perception [BRM], [CA], [MAH], [DB]

71.Gv Measures of speech perception (intelligibility and quality) [VB], [ICB], [CGC], [MAH], [BRM], [MAS]

71.Hw Cross-language perception of speech [BRM], [MAH], [CGC]

71.Ky Speech perception by the hearing impaired [DB], [VB], [FJG], [ICB], [PBN], [EB]

71.Lz Speech perception by the aging [DB], [PBN], [MAH]

71.Qr Neurophysiology of speech perception [ICB], [MAH]

71.Rt Sensory mechanisms in speech perception [ICB], [MAH], [DB]

71.Sy Spoken language processing by humans [DB], [MSV], [MAH], [CGC]

[72] Speech processing and communication systems

72.Ar Speech analysis and analysis techniques; parametric representation of speech [CYE], [SSN], [SAF]

72.Bs Neural networks for speech recognition [CYE], [SSN]

72.Ct Acoustical methods for determining vocal tract shapes [CYE], [SSN], [ZZ]

72.Dv Speech-noise interaction [CYE], [SSN]

72.Fx Talker identification and adaptation algorithms [CYE], [SSN], [SAF]

72.Gy Narrow, medium, and wideband speech coding [CYE], [SSN]

72.Ja Speech synthesis and synthesis techniques [CYE], [SSN], [SAF]

72.Kb Speech communication systems and dialog systems [CYE], [SSN]

72.Lc Time and frequency alignment procedures for speech [CYE], [SSN]

72.Ne Automatic speech recognition systems [CYE], [SSN], [SAF]

72.Pf Automatic talker recognition systems [CYE], [SSN], [SAF]

72.Qr Auditory synthesis and recognition [CYE], [SSN], [SAF]

72.Uv Forensic acoustics [CYE], [SAF]

[75] Music and musical instruments

75.Bc Scales, intonation, vibrato, composition [DD], [MAH]

75.Cd Music perception and cognition [DD], [MAH], [DB]

75.De Bowed stringed instruments [TRM], [JW]

75.Ef Woodwinds [TRM], [JW], [AH]

75.Fg Brass instruments and other lip vibrated instruments [TRM], [JW], [ZZ]

75.Gh Plucked stringed instruments [TRM], [JW]

75.Hi Drums [TRM]

75.Kk Bells, gongs, cymbals, mallet percussion and similar instruments [TRM]

75.Lm Free reed instruments [TRM], [JW], [AH], [ZZ]

75.Mn Pianos and other struck stringed instruments [TRM]

75.Np Pipe organs [TRM], [JW]

75.Pq Reed woodwind instruments [AH], [TRM], [JW], [ZZ]

75.Qr Flutes and similar instruments [AH], [TRM], [JW]

75.Rs Singing [DD], [TRM], [JW]

75.St Musical performance, training, and analysis [DD], [DB]

75.Tv Electroacoustic and electronic instruments [DD]

75.Wx Electronic and computer music [MAH]

75.Xz Automatic music recognition, classification and information retrieval [DD], [SSN]

75.Yy Instrumentation measurement methods for musical acoustics [TRM], [JW]

75.Zz Analysis, synthesis, and processing of musical sounds [DD], [MAH]

[80] Bioacoustics

80.Cs Acoustical characteristics of biological media: molecular species, cellular level tissues [RRF], [MJO], [DLM], [TK], [SWY], [GH], [KAW], [TK]

80.Ev Acoustical measurement methods in biological systems and media [DLM], [RRF], [MJO], [SWY], [GH], [KAW]

80.Gx Mechanisms of action of acoustic energy on biological systems: physical processes, sites of action [ANP], [RRF], [MJO], [GH], [SWY], [KAW]

80.Jz Use of acoustic energy (with or without other forms) in studies of structure and function of biological systems [ANP], [RRF], [MJO], [DLM], [GH], [SWY], [KAW]

80.Ka Sound production by animals: mechanisms, characteristics, populations, biosonar [AMS], [ANP], [DKM], [JJF], [MJO], [AMT], [ZZ]

80.Lb Sound reception by animals: anatomy, physiology, auditory capacities, processing [AMS], [ANP], [DKM], [JJF], [MJO]

80.Nd Effects of noise on animals and associated behavior, protective mechanisms [AMS], [ANP], [DKM], [JJF], [MJO], [AMT]

80.Pe Agroacoustics [RRF], [WA], [MCH]

80.Qf Medical diagnosis with acoustics [MDV], [DLM], [GH], [SWY], [KAW]

80.Sh Medical use of ultrasonics for tissue modification (permanent and temporary) [DLM], [ROC], [MDV], [GH], [SWY], [KAW]

80.Vj Acoustical medical instrumentation and measurement techniques [DLM], [MCH], [MDV], [GH], [SWY], [KAW]

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, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502. Telephone: (516) 576-2360; E-mail: asa@aip.org

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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.

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____

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Size of Business: [ ] Small business [ ] Small business—No Journal [ ] Large business

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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:

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It is understood that a Sustaining Member will not use the membership for promotional purposes.

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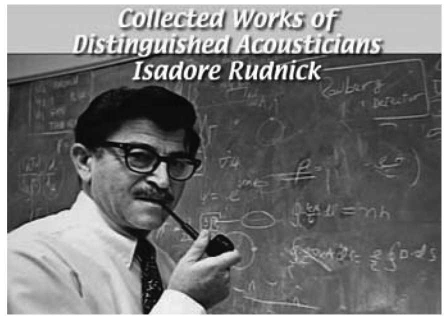
Please send completed applications to: Executive Director, Acoustical Society of America, Suite 1NO1, 2 Huntington Quadrangle, Melville, NY 11747-4502, (516) 576-2360

Collected Works of Distinguished Acousticians

Isadore Rudnick



The first in this series of the Collected Works of Distinguished Acousticians is that of Isadore Rudnick (May 8, 1917 - August 22, 1997). Rudnick was honored by the Acoustical Society of America (ASA) with the R. Bruce Lindsay (Biennial) Award in 1948, the Silver Medal in Physical Acoustics in 1975, and the Gold Medal in 1982. He was recognized for his acoustics research in low temperature physics with this field's most prestigious award, the Fritz London Memorial Award, in 1981 and was inducted into the National Academy of Science in 1983. Izzy's research in physical acoustics addressed boundary propagation, reciprocity calibration, high intensity sound and its biological effects, nonlinear sound propagation, and acoustics in superconductors and superfluids, including critical phenomena in bulk and thin films. The first disc in this three disc set contains reprints of Rudnick's papers from scientific journals, including 26 from the Journal of the Acoustical Society of America, and 87 from other prestigious journals, as well as some consulting reports and invited papers presented at international meetings which would otherwise be difficult to obtain. The second disc includes a montage of photographs of Rudnick with colleagues and family, Rudnick's prize winning film "The Unusual Properties of Liquid Helium", and a video of the Plenary session at the ASA's 100th meeting where Rudnick presented 90 minutes of unique and stage-sized acoustics demonstrations. While videotaped under poor conditions and of lamentable quality, the reprocessed video of acoustics demonstrations is one of the most valuable parts of this collection. The third disc is a video recording of the Memorial Session held at the 135th meeting of the ASA, which provides a comprehensive summary of Rudnick's contributions as described by former students and collaborators.

The CD was compiled by Julian D. Maynard and Steven L. Garrett of the Pennsylvania State University, State College, Pennsylvania.

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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 On-line–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	*	*	On-line	*
Meeting Calls for Papers	On-line	On-line	On-line	On-line
Reduced Meeting Registration Fees	*	*		*
5 Free ASA standards per year-download only	*	*		*
Standards Discounts	*	*		*
Society Membership Directory	On-line	On-line	On-line	On-line
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* and its supplements (meeting programs). **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.**
- CD-ROM features: full-text searching with support for advanced searching (boolean, proximity, soundex, etc.); easy on-screen reading; excellent print quality; only the final (year-end) CD-ROM need be retained; uses the popular Adobe Acrobat Reader; works on any standard pc capable of running Windows and any Macintosh running System 7 or higher; can also be used on Unix X-window systems (full-text searching is not supported on Unix version at this time); toll-free (U.S.) technical support provided.
- COMBINATION OF THE CD-ROM AND PRINTED JOURNAL. The CD-ROM mailed bimonthly and the printed journal mailed monthly. This is for those members who wish to receive the complete printed Journal each month, and to have the CD-ROM for retention and search purposes. **Cost: Add $70 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 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 $160 in addition to dues.
JASA on CD-ROM is sent by air mail at no charge in addition to dues.
Note: Dues payment includes a non-deductible allocation of $5.00 for Acoustics Today.

ACOUSTICAL SOCIETY OF AMERICA

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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 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 Journals subscription 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 $160. 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
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CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
NAME OF ORGANIZATION OR BUSINESS			
DEPARTMENT			
ORGANIZATION ADDRESS (STREET & NUMBER)			
CITY	STATE OR PROVINCE	ZIP OR POSTAL CODE	COUNTRY
BUSINESS TELEPHONE: AREA CODE/NUMBER	FAX: AREA CODE/NUMBER	HOME TELEPHONE: AREA CODE/NUMBER	
E-MAIL ADDRESS: (PRINT CLEARLY)			
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**
- ☐ BIOMEDICAL ACOUSTICS **K**
- ☐ NOISE & NOISE CONTROL **D**
- ☐ SIGNAL PROCESSING IN ACOUSTICS **N**
- ☐ ANIMAL BIOACOUSTICS **L**
- ☐ ENGINEERING ACOUSTICS **B**
- ☐ PHYSICAL ACOUSTICS **E**
- ☐ SPEECH COMMUNICATION **H**
- ☐ STRUCTURAL ACOUSTICS & VIBRATION **G**
- ☐ ARCHITECTURAL ACOUSTICS **A**
- ☐ MUSICAL ACOUSTICS **C**
- ☐ PSYCHOLOGICAL & PHYSIOLOGICAL ACOUSTICS **F**
- ☐ 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, SUITE 1NO1, 2 HUNTINGTON QUADRANGLE, MELVILLE, NY 11747-4502.

METHOD OF PAYMENT

☐ Check or money order enclosed for $ ____________ (U.S. funds/drawn on U.S. bank)

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(Credit card orders must be signed)

Account Number ☐☐☐☐☐☐☐☐☐☐☐☐☐☐☐☐

Expiration Date Mo. ☐☐ Yr. ☐☐

Security Code ☐☐☐☐

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: Robert C. Coffeen, University of Kansas, Lawrence, KS 60045, coffeen@ku.edu. Catherine Rogers, University of South Florida, Tampa, FL, crogers@cas.usf.edu

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Campanella Associates
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Email: a.campanella@att.net

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Columbia College Chicago
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Chicago, IL 60605
Email: lronsse@colum.edu

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Dept. of Audio Arts and Acoustics
Columbia College Chicago
33 E. Congress Pkwy., Rm. 6010
Chicago, IL 60605
Email: sguzman@colum.edu

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Richard J. Morris
Communication Science and Disorders
Florida State Univ.
201 W. Bloxham
Tallahassee, FL 32306-1200
Email: richard.morris@cci.fsu.edu

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Reuter Associates, LLC
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Tel: 603-430-2081
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Tel: 972-883-2435
Email: assmann@utdallas.edu

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Victoria Suha
Email: suha.v@husky.neu.ed

ORANGE COUNTY
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Email: dlubman@dlacoustics.com

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Anand Swaminathan
Pennsylvania State Univ.
201 Applied Science Bldg.
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Purdue Univ.
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Tel: 765-494-1099
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SPAWAR Systems Center, Pacific
49575 Gate Road, Room 170
San Diego, CA 92152-6435
Tel: 619-553-5634
Email: paul.baxley@navy.mil

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Camilo Perez
Applied Physics Lab.
Univ. of Washington
1013 N.E. 40th St,
Seattle, WA 98105-6698
Email: camipiri@uw.edu

UPPER MIDWEST

David Braslau
David Braslau Associates, Inc.
6603 Queen Ave. South, Ste. N
Richfield, MN 55423
Tel: 612-331-4571
Email: david@braslau.com

WASHINGTON, DC

Matthew V. Golden
Scantek, Inc.
6430 Dobbin Rd., Ste. C
Columbia, MD 21045
Tel: 410-290-7726
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ACOUSTICAL SOCIETY OF AMERICA
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ACOUSTICAL DESIGN OF MUSIC EDUCATION FACILITIES. Edward R. McCue and Richard H. Talaske, Eds. Plans, photographs, and descriptions of 50 facilities with explanatory text and essays on the design process. 236 pp, paper, 1990. Price: $23. **Item # 0-88318-8104**

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