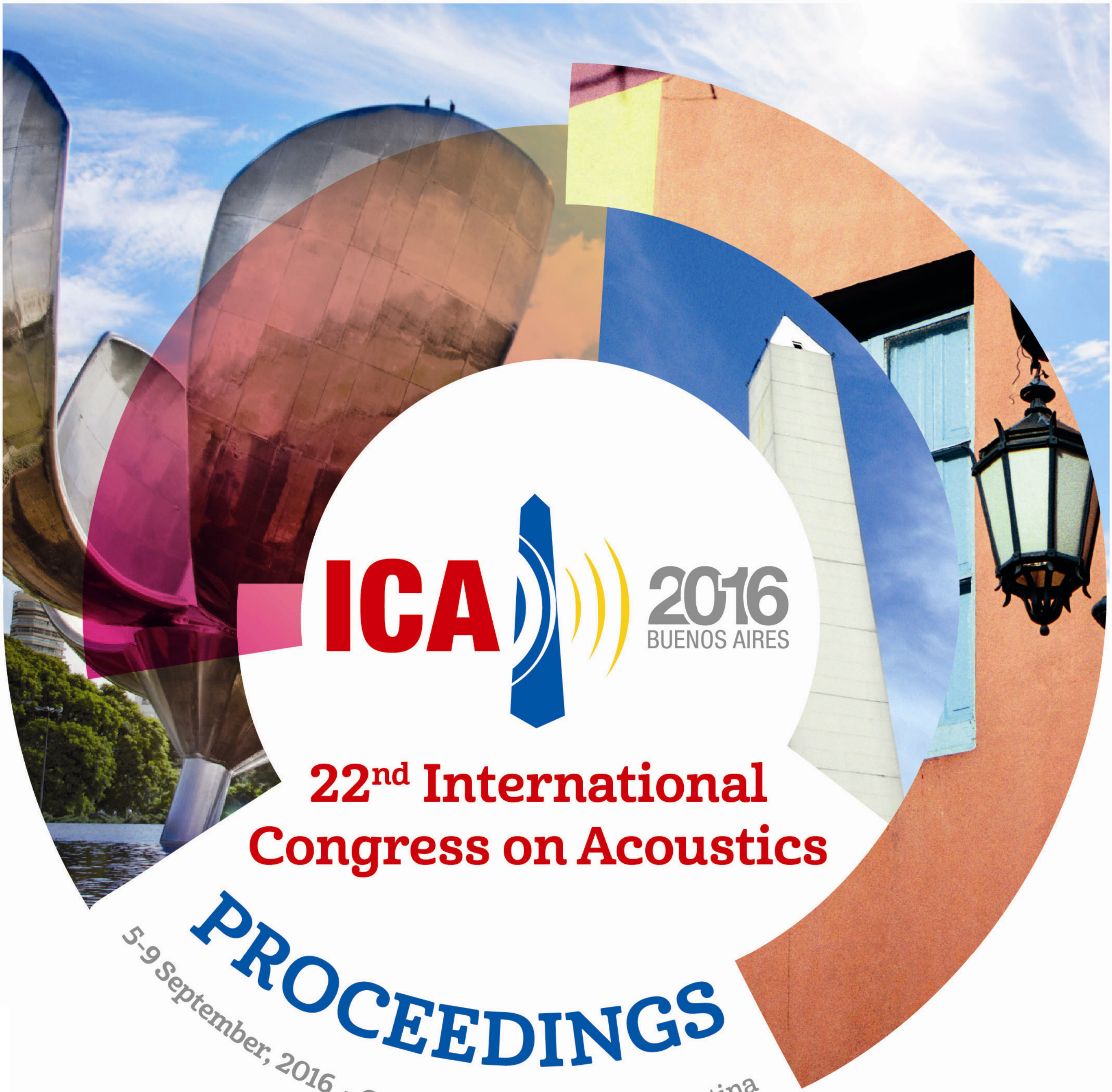

ICA 2016 BUENOS AIRES

22nd International Congress on Acoustics

PROCEEDINGS

5-9 September, 2016 - Catholic University of Argentina

X Congreso Iberoamericano de Acústica

XIV Congreso Argentino de Acústica

XXVI Encontro da Sociedade Brasileira de Acústica

Editors:

. Federico Miyara | . Vivian Pasch
. Ernesto Accolti | . Nilda Vechiatti

Asociación de Acústicos Argentinos | Federación Iberoamericana de Acústica | International Commission for Acoustics | SOBRAC Sociedade Brasileira de Acústica www.acustica.org.br

PROCEEDINGS OF THE ICA CONGRESS (online) - ISSN 2415-1599

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22nd International Congress on Acoustics ICA 2016

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Proceedings of the
22th International Congress on Acoustics
ICA 2016

5-9 September 2016
Catholic University of Argentina, Buenos Aires, Argentina

ICA 2016 has been organised by the Ibero-american Federation of Acoustics (FIA) and the Argentinian Acousticians Association (AdAA) on behalf of the International Commission for Acoustics.

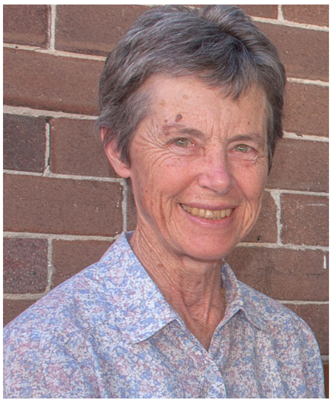
The program incorporates the X Ibero-american Federation of Acoustics Congress, the XIV Argentine Congress of Acoustics, and the XXVI Meeting of the Brazilian Acoustical Society

The purpose of the ICA is to promote international development and collaboration in all fields of acoustics including research, development, education, and standardisation. It fulfils this mission by maintaining contacts with national and regional acoustical societies and associations; convening the International Congresses on Acoustics; sponsoring special international meetings. The Commission is affiliated to the International Union for Pure and Applied Physics; the International Union of Applied and Theoretical Mechanics and the International Council of Scientific Unions.

The Ibero-American Federation of Acoustics (Federación Iberoamericana de Acústica – FIA) is a non-profit scientific association, created in October 1995, in Valdivia –Chile, being its members the Acoustical Societies of the Spanish and Portuguese speaking countries. It seeks to to establish links between acoustics associations, private and public companies, scientific institutes and universities, syndicates, etc. To this end it organizes congresses, workshops, courses and meetings and promotes cooperation agreements

The Argentinian Acousticians Association (AdAA) is a non-profit academic organization that aims to raise awareness of the worth and utility of Acoustics in its many areas and applications by bringing together scientists, professionals, interested people and organizations. To this end it encourages research and exchange of knowledge in all fields of acoustics, organizes meetings and congresses as well as activities for the general public such as the Week of Sound. The Society was founded in 1976, so this year it celebrates its 40th anniversary.

Welcome Message from Marion Burgess President, International Commission of Acoustics, 2013-2016



It is with great pleasure that on behalf of the International Commission for Acoustics I welcome you to the ICA 2016 in Buenos Aires. The selection of the hosts for the congress of the ICA is undertaken 6 years before the actual event and since that time there have been three Presidents of the ICA. However the organising committee has worked throughout this time to bring to fruition this congress. That committee comprised the Ibero-american Federation of Acoustics (FIA) and the Argentinian Acousticians Association (AdAA), in cooperation with the Chilean Acoustics Society (SOCHA). We thank the efforts of the committee for working on this congress over so many years and in particular thank the Congress President, Jorge Patricio and the Congress Secretary General Nilda Vechiatti.

The International Congress on Acoustics provides the opportunity only once every three years for all those around the world who are working in all areas of acoustics to meet, discuss and exchange ideas. We hope that during the congress you will benefit from the opportunity for cross disciplinary discussion. The five plenary lectures provide insights into five very different areas of acoustics. This highlights the breadth of the coverage of the presentations at the congress. The main program involves the presentations in a range of special topic areas and structured sessions. The technical exhibition provides the opportunity to see the latest advances in instrumentation and products related to acoustics.

The ICA provides grants to support the participation of some Young Scientists at the congress. We are delighted that additional funding from the International Union for Pure and Applied Physics (IUPAP) has been used to extend the Young Scientists support scheme.

This is the first time that the ICA has been held in South America so it is a wonderful opportunity for those from other parts of the world to learn more about the work being undertaken in these countries. It also provides the opportunity to learn more about the cultural and social life of Argentina and, perhaps for some, the opportunity to also travel to the nearby countries.

Thank you to all those who have worked and continue to work on the arrangements for this congress and, equally importantly, thank you to those who are participating and presenting their work to colleagues.

Welcome Message from Jorge Patricio President, Iberomaerican Federation of Acoustics, President, International Congress on Acoustics

It is with great pleasure that I welcome you all to the 22nd International Congress on Acoustics, ICA 2016, that is being held in Buenos Aires, Argentina, between 5th and 9th, September, 2016.

This congress is one of the most important forums for those around the world working in all areas of Acoustics. It makes an outstanding opportunity to meet, discuss, exchange ideas, and establish partnerships, thus promoting the research and development of this scientific branch of Knowledge, in order to duly serve the needs of the Human being.

Jointly, with the ICA2016, it will also be held the X Ibero-american Federation of Acoustics Congress (FIA 2016), incorporating the XIV Argentinean Congress of Acoustics and the XXVI Meeting of the Brazilian Acoustical Society.

The ICA 2016 Congress is organized by the Ibero-american Federation of Acoustics (FIA) and the Argentinian Acousticians Association (AdAA), in cooperation with the Chilean Acoustics Society (SOCHA), under the endorsement of the International Commission for Acoustics (ICA). It is officially sponsored by the Acoustical Society of America (ASA), the International Union of Pure and Applied Physics (IUPAP), and the National Council of Acoustical Consultants (NCAC), Institutions to whom I express my deepest gratitude.

Although we are currently suffering strange and unexpected turbulences worldwide, Acoustics still remains as an area of scientific and technical knowledge with strong dynamism. That is evident in the almost 600 papers to be presented in the frame of the Congress, among which 5 plenary lectures deserve the utmost attention since they deal with the latest developments in emergent scientific fields. A technical exhibition of materials, products, equipment and services of innovative character, presented by several companies to whom I thank for their participation and interest, will also be a key mark of the Congress.

Within the ICA umbrella, two short courses are planned to occur. Also, as satellite event, a Symposium on Room and Musical Acoustics will be held immediately after ICA 2016, in the Argentinian city of La Plata.

The accompanying persons programme, scheduled for these days, and the various social activities, turn this attendance into an excellent and unique opportunity to enjoy the Congress, and a chance to visit the magnificent and astonishing city of Buenos Aires, the Paris of South America with its culture and history, being proud to host their visitors with hospitality and a smile of hope in the future.

Finally, and on behalf of the organizing team of ICA 2016, I want to stress my deepest thanks for your participation, as well as for the involvement of all supporting institutions and persons with a special mention to Prof. Nilda Vechiatti (Secretary General of ICA 2016) for her dedication, commitment and work.

So, on behalf of ICA 2016 organization and the Ibero-american Federation of Acoustics, I wish you all an excellent stay in Buenos Aires, in Argentina and in South America.

Welcome Message from Nilda Vechiatti President, Argentinian Acousticians Association, 2012-2016

As the host of this 22nd International Congress on Acoustics ICA 2016, we are delighted to welcome everyone here on behalf of the acousticians of Latin America and, in particular, of Argentina.

I would like to thank Jorge Patricio because without his leadership and experience it would not have been possible to organize the ICA 2016 Congress.

I would also like to thank Marion Burgess, Mike Stinson and Michael Vorländer, the Executive Board of ICA, for their support and assistance with the organization (with special emphasis on Marion's contribution).

Thanks to the people of MCI as well, for their professional experience at the service of the organization of our event.

We hope that this Congress, held for the first time in Latin America will trigger further development of Acoustics and its applications in the region, and enhance our links to the acoustical community around the world.

It is also a great honor for us to incorporate the 10th Congress of the Ibero-american Federation of acoustics, the 26th Meeting of the Brazilian Acoustical Society and the 14th Argentinian Congress of Acoustics within this ICA Congress, especially in this year when the Argentinian Acousticians Association celebrates its first 40 years. We are persuaded that these events will be a bridge to reduce the distance among the acousticians of these countries.

I am pleased to announce that with the support of the International Union of Pure and Applied Physics, we have offered partial scholarships to facilitate the attendance of acousticians from developing countries, and particularly from Latin America countries.

We heartily wish you a very pleasant stay in Buenos Aires and Argentina, and also that you enjoy the congress.

Thank you!

INTERNATIONAL ORGANISING COMMITTEE

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Secretary General	Nilda Vechiatti (Argentina)
Technical Chairs	Samir Gerges (Brasil) Pablo Girón (Argentina)
Technical Program	Purificación Merodo (Argentina) Jaime Delannoy (Chile)
Treasurer	Daniel Muzzio (Argentina)

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Technical Program	Purificación Merodo Adrián Azurro
Proceedings	Federico Miyara Ernesto Accolti Vivian Pasch Nilda Vechiatti
Social Program	Vivian Pasch
Exposition	Daniel Muzzio Alejandro Giani
Sponsorship	Walter Montano Pablo Ciccarella

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Dinara Xavier da Paixão	Brazil
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Kohei Yamamoto	Japan

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Cesar Díaz Sanchidrian	Luc Mongeau	Masuzo Yanagida
Bertrand Dubus	Jorge Moreno Ruiz	

Technical Program
Oral Sessions

Monday, September 5							
Room	Juan Pablo II Auditorium	Dr. Valsecchi Auditorium	Cardenal Pironio Auditorium	Room 204	Micro cinema	Auditorium 2	Auditorium 3
09:00 - 11:00	**REGISTRATION PROCESS** San José building, South access, Ground floor						
11:00 - 12:00	**OPENING CEREMONY** Juan Pablo II Auditorium, 2nd floor						
12:00 - 13:00	Plenary Lecture: **Michael Vorländer** **"FROM ACOUSTIC SIMULATION TO VIRTUAL AUDITORY DISPLAYS"** Juan Pablo II Auditorium, 2nd floor						
13:00 - 14:30	**BREAK**						
14:30 - 16:10	EN2-628	BA1-1135	AA5-621	SC1-420	AB1-96	NT1-864	PA1-74
	EN2-401	BA1-148	AA5-658	SC1-586	AB1-908	NT1-19	PA1-579
	EN2-517	BA1-563	AA5-50	SC1-699	AB1-772	NT1-181	PA1-104
	EN2-458	BA1-284	AA5-121	SC1-714	AB1-866	NT1-309	PA1-368
	EN2-739	BA1-405	AA5-483	SC1-662	AB1-820	NT1-418	PA1-61
16:10 - 16:30	**BREAK**						
16:30 - 18:30	EN1-723	BA1-630	AA5-89	SC1-333	PP2-672	NT2-802	PA1-620
	EN1-410	BA1-642	AA5-337	SC1-357	PP2-557	NT2-101	PA1-345
	EN1-378	BA1-643	AA5-379	SC1-721	PP2-113	NT2-561	PA1-402
	EN1-218	BA1-646	AA5-381	SC1-847	PP2-558	NT2-736	PA1-896
	EN1-216	BA1-647	AA5-843	SC1-722	PP2-907	NT2-210	
	EN1-157	BA1-651	AA5-463			NT2-426	
18:30 - 19:30	**WELCOME COCKTAIL** Foyer of Juan Pablo II Auditorium, 2nd floor						

Technical Program
Oral Sessions

Tuesday, September 6							
Room	Juan Pablo II Auditorium	Dr. Valsecchi Auditorium	Cardenal Pironio Auditorium	Room 204	Micro cinema	Auditorium 2	Auditorium 3
09:00 - 10:40	EN1-092	BA1-649	AA4-132	AA8-180	FIA-NS-6		
	EN1-067	BA1-789	AA4-372	AA8-301	FIA-NS-100		PA2-20
	EN1-666	BA1-804	AA4-271	AA8-487	FIA-NS-116	NT2-336	PA2-202
	EN1-056	BA1-725	AA4-255	AA8-76	FIA-NS-102	NT2-362	PA2-59
	EN1-081	BA1-364	AA4-159	AA8-716	FIA-PP-37	NT2-371	PA2-107
1040 - 11:00	BREAK						
11:00 - 12:00	EN1-829	AA5-55	AA4-163	AA8-811	FIA-EN-86	NT2-523	PA2-374
	EN1-254	AA5-765	AA4-165	AA8-160	FIA-EN-26	NT2-118	PA2-90
	EN1-450	AA5-325	AA4-683	AA8-861	FIA-EN-28	NT2-133	PA2-580
12:00 - 13:00	Plenary Lecture: **Chen-Fen Huang** **"ON THE PERSPECTIVE OF UNDERWATER ACOUSTIC TOMOGRAPHY FOR PROBING OCEAN CURRENTS IN SHALLOW-WATER ENVIRONMENTS"** Juan Pablo II Auditorium, 2nd floor						
13:00 - 14:30	BREAK						
14:30 - 16:10	EN1-105	AA5-169		SP2-489	FIA-EN-29	CA2-187	PA3-543
	EN1-247	AA5-238		SP2-501	FIA-EN-49	CA2-790	PA3-48
	EN1-390	AA5-205		SP2-567	FIA-EN-12	CA2-109	PA3-595
	EN1-429	AA5-809	SV1-654	SP2-603	FIA-EN-88	CA2-391	PA3-26
	EN1-901	AA5-822	SV1-857	SP2-607	FIA-EN-103	CA2-496	PA3-29
16:10 - 16:30	BREAK						
	Environmental Protection Agency, Ministry of Environment and Public Space: "STRATEGIC NOISE MAP OF THE CITY OF BUENOS AIRES" Cardenal Pironio Auditorium, 1st floor						
16:30 - 18:50	EN1-079	AA5-552	SV2-44	SP2-813	FIA-EN-43	CA2-465	PA3-94
	EN1-629	AA5-484	SV2-78	SP3-272	FIA-EN-73	CA2-318	PA3-103
	EN1-417	AA5-21	SV2-80	SP3-141	FIA-EN-84	CA2-43	PA3-174
	EN1-657	AA5-831	SV2-10	SP4-111	FIA-EN-98	CA2-818	PA3-175
		AA5-224	SV2-41	SP4-728	FIA-EN-11	CA2-317	PA3-251
		AA5-537	SV2-176	SP4-608	FIA-EN-47		PA3-264
		AA5-451	SV2-602	SP4-581	FIA-EN-75		PA3-296

Technical Program
Oral Sessions

Wednesday, September 7							
Room	Juan Pablo II Auditorium	Dr. Valsecchi Auditorium	Cardenal Pironio Auditorium	Room 204	Micro cinema	Auditorium 2	Auditorium 3
09:00 - 10:40	**EN5-245**	**SP1-731**	**AA1-178**	**SV2-182**	**FIA-AA-52**		**UW1-780**
	EN5-070	**SP1-158**	**AA1-241**	**SV2-99**	**FIA-AA-72**		**UW1-917**
	EN5-223	**SP1-204**	**AA1-524**	**SV2-387**	**FIA-AA-93**		**UW1-427**
	EN5-86	**SP1-312**	**AA1-278**	**SV2-516**	**FIA-AA-51**	**AA2-124**	**UW1-438**
	EN5-168	**SP1-412**	**AA1-828**	**SV2-570**	**FIA-AA-67**	**AA2-213**	**UW1-834**
1040 - 11:00	**BREAK**						
11:00 - 12:00	**EN5-239**	**SP1-122**	**ED1-724**	**SV2-618**	**FIA-AA-45**	**AA2-269**	**UW1-636**
	EN5-353	**SP1-139**	**ED1-83**	**SV2-798**	**FIA-SS-122**	**AA2-77**	**UW1-138**
	EN5-673	**SP1-593**	**ED1-763**	**SV2-824**	**FIA-SS-111**	**AA2-591**	**UW1-644**
12:00 - 13:00	Plenary Lecture: **Frank Russo** **"UNDERSTANDING MUSIC PERCEPTION FROM THE PERSPECTIVE OF OSCILLATION AND RESONANCE"** Juan Pablo II Auditorium, 2nd floor						
13:00 - 14:30	**BREAK**						
	SIMPOSYUM GINER-CDM **"Insulation and acoustic conditioning project of all the technical rooms of the new Disney Latin America headquarters in Argentina"** Cardenal Pironio Auditorium, 1st floor						
14:30 - 16:10	**EN5-316**	**SP1-598**	**AA1-830**	**PA3-305**	**FIA-SP-35**		**UW1-262**
	EN5-440	**SP1-686**	**AA1-585**	**PA3-386**	**FIA-SP-53**		**UW1-398**
	EN5-444	**SP1-415**	**AA1-817**	**PA3-531**	**FIA-MU-22**		**UW1-35**
	EN5-565	**SP1-903**	**AA1-102**	**PA3-741**	**FIA-MU-96**		**UW1-164**
	EN5-503	**SP1-506**	**AA1-268**	**PA3-943**	**FIA-MU-115**		**UW1-446**
16:10 - 16:30	**BREAK**						
16:30 - 18:30	**ICA GENERAL ASSEMBLY** Cardenal Pironio Auditorium, 1st floor						

Technical Program
Oral Sessions

Thursday, September 8							
Room	Juan Pablo II Auditorium	Dr. Valsecchi Auditorium	Cardenal Pironio Auditorium	Room 204	Micro cinema	Auditorium 2	Auditorium 3
09:00 - 10:40	EN4-97	SS4-226	AA6-606	CA1-363	Course: "Acoustic Design of Mufflers"		Course: "Ultrasound, Cavitation, Sonochemistry"
	EN4-330	SS4-233	AA6-166	CA1-715			
	EN4-545	SS4-265	AA6-729	CA1-827			
	EN4793	SS4-340	AA6-95	CA1-718			
		SS4-167	AA6-664	CA1-633			
1040 - 11:00	BREAK						
11:00 - 12:00	EN3-850	SI1-208	AA6-732	CA1-299	"Acoustic Design of Mufflers"	PP3-885	"Ultrasound, Cavitation, Sonochemistry"
	EN3-564	SI1-534	AA6-32	CA1-154		PP3-219	
	EN3-904	SI1-486	AA6-754	CA1-346		PP3-832	
12:00 - 13:00	Plenary Lecture: **Barbara Shinn-Cunningham** **"HOW THE BRAIN MAKES SENSE OF COMPLEX AUDITORY SCENES"** Juan Pablo II Auditorium, 2nd floor						
13:00 - 14:30	BREAK						
14:30 - 16:10	NS2-98	SI1-198	AA6-1182	CA1-445	MU1-243	PP3-240	VA1-519
	NS2-808	SI2-859	AA6-743	CA1-253	MU1-227	PP3-432	VA1-768
	NS2-354	SI2-792	AA6-320	CA1-161	MU1-152	PP3-435	VA1-886
	NS3-495	SI2-388	AA6-761	CA1-525	MU1-727	PP3-288	VA1-797
	NS3-684	SI2-190			MU1-1179	PP3-9	VA1-801
16:10 - 16:30	BREAK						
16:30 - 18:50	NS4-535	SI2-87	SS5-746	AO1-687	MU5-858	PP3-730	VA1-681
	NS4-597	SI2-635	SS5-706	AO1-361	MU5-750	PP3-612	VA1-170
	NS4-491	SI2-382	SS5-674	AO1-298	MU5-692	PP3-766	VA1-69
	NS4-490	SI2-619	SS2-678	AO1-360	MU5-601	PP3-150	VA1-826
	NS4-589	SI2-548	SS2-656	AO1-338	MU5-207	PP3-665	
	NS4-882	SI2-421		AO1-196	MU5-482	PP3-279	
16:30 h - Tecnical Visit to Disney Latin America facilities, for previously registered participants LIMITED CAPACITY (event not organized by the ICA organizers) Sponsored by Giner							

Technical Program
Oral Sessions

Friday, September 9							
Room	Juan Pablo II Auditorium	Dr. Valsecchi Auditorium	Cardenal Pironio Auditorium	Room 204	Micro cinema	Auditorium 2	Auditorium 3
09:00 - 10:40	AA7-863	SS3-550	NS4-911				
	AA7-578	SS3-528	NS4-323	EL1-290	MU3-54	NS5-605	
	AA7-527	SS3-856	NS1-110	EL1-436	MU3-399	NS5-188	AA3-713
	AA7-624	SS3-532	NS1-93	EL1-655	MU3-459	NS5-852	AA3-383
	AA7-556	SS3-791	NS1-214	EL1-355	MU3-560	NS5-201	AA3-464
1040 - 11:00	**BREAK**						
11:00 - 13:00	AA7-703	SS3-609	NS1-443	EL1-810	MU4-748		AA3-156
	AA7-777	SS3-222	NS1-131	EL1-215	MU4-756	PP1-711	AA3-127
	AA7-717	SS3-720	NS1-108	EL1-155	MU4-610	PP1-653	AA3-197
	AA7-919	SS3-263	NS1-430	EL1-193	MU4-347	PP1-431	AA3-900
	AA7-277	SS3-119	NS1-191	EL1-147	MU4-870	PP1-776	AA3-313
	AA7-225	SS3-75		EL1-47		PP1-123	AA3-572
13:00 - 14:30	**BREAK**						
14:30 - 15:50	AA7-232	SS3-267	NS6-116	EL1-230	MU2-8	PP1-493	
	AA7-551	SS3-794	NS6-276	EL1-422	MU2-25	PP1-775	
	AA7-574		NS6-773	EL1-559	MU2-171	PP1-373	
	AA7-393			EL1-613		PP1-400	
15:50 - 16:50	Plenary Lecture: **Samir Gerges** **"HEARING PROTECTORS: STATE OF THE ART AND EMERGING TECHNOLOGIES OF COMFORT AND UNCERTAINTY IN MEASUREMENTS"** Juan Pablo II Auditorium, 2nd floor						
16:50 - 17:30	**CLOSING CEREMONY** Juan Pablo II Auditorium, 2nd floor						
17:30 - 18:30	**FAREWELL COCKTAIL** Foyer of Juan Pablo II Auditorium, 2nd floor						

Technical Program
Poster Sessions

Tuesday, September 6				Wednesday, September 7				Thursday, September 8			
E-POSTERS SESSION Lounge lateral room				E-POSTERS SESSION Lounge lateral room				E-POSTERS SESSION Lounge lateral room			
	LCD 1	LCD 2	LCD 3		LCD 1	LCD 2	LCD 3		LCD 1	LCD 2	LCD 3
09:40	AA5-57	NS1-615	CA2-833	09:40		PA2-922	PA3-521	09:40			FIA-PP-95
10:00	AA5-442	NS5-695	SP1-295	10:00	NT2-45	PA3-406	SC1-786	10:00	MU2-821		FIA-PP-79
10:20	AA3-409	NS5-865	SP1-352	10:20	NT2-668	PA3-221	SC1-185	10:20	MU3-576	VA1-342	FIA-SS-60
10:40	BREAK			10:40	BREAK			10:40	BREAK		
11:00	AA3-707	NS4-128	SP3-367	11:00	AA2-331	PA3-344	SC1-738	11:00	PP1-53	SP4-125	FIA-NS-64
11:20	AA3-704	NS4-326	SP3-530	11:20	ED1-366	PA3-348	SC1-300	11:20	PP1-816	SP4-303	FIA-NS-17
11:40	AA3-162	NS4-475		11:40	ED1-477	PA3-819	SC1-526	11:40	PP2-851	SP4-304	FIA-NS-27
12:00	Plenary Lecture			12:00	Plenary Lecture			12:00	Plenary Lecture		
13:00				13:00				13:00			
14:30				14:30				14:30			
14:50				14:50	AA6-676	AO1-518	SC1-645	14:50		SP4-306	FIA-EN-30
15:10	AA2-815	BA1-292		15:10	AA6-511	UW1-151	SC1-140	15:10	PP3-114	SP4-307	FIA-EN-34
15:30	SS3-329	BA1-324	SI1-533	15:30	AA6-502	UW1-192	SC1-544	15:30	PP3-252	SP4-308	FIA-EN-61
15:50	SS3-476	BA1-499	SI2-784	15:50	AA6-694	UW1-235	SC1-770	15:50	PP3-582	SP4-310	FIA-EN-62
16:10	BREAK			16:10	BREAK			16:10	BREAK		
16:30	EN1-229	NS4-604	EL1-747	16:30				16:30	PP3-625	SP4-314	FIA-AA-56
16:50	EN2-478	NS4-660	EL1-425	16:50				16:50	PP3-700	SP4-614	FIA-AA-76
17:10	EN3-321	NS4-701	EL1-488	17:10				17:10	PP3-735	SP4-881	FIA-AA-70
17:30				17:30				17:30			FIA-AA-38
17:50				17:50				17:50			FIA-AA-46
18:10				18:10				18:10			FIA-AA-55

ICA 2016 Topics
Subtopics (Structured Sessions)

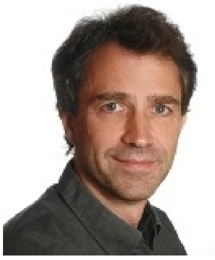
ICA 2016 CONGRESS TOPICS	SUBTOPICS (STRUCTURED SESSIONS)
AA - Architectural Acoustics - Room and Building Acoustics	**AA1 - Acoustics In Education**
	AA2 - Acoustic Of Worship Spaces
	AA3 - Architectural Acoustics for Non-Performance Spaces
	AA4 - Calculation Models for Timber Structures (Silent Timber Build)
	AA5 - Challenges and Solutions in Acoustics Measurements and Design
	AA6 - Concert Hall Acoustics
	AA7 - Isotropy and Diffuseness in Room Acoustics
	AA8 - Legislation and Regulations in Building Acoustics
AB - Animal Bioacoustics	**AB1 - Animal Bioacoustics**
AO - Acoustical Oceanography	**AO1 - Acoustical Oceanography**
BA - Biomedical Acoustics	**BA1 - Biomedical Acoustics**
CA - Communication Acoustics	**CA1 - The Technology of Binaural Listening & Understanding**
	CA2 - Communication Acoustics
ED - Education in Acoustics	**ED1 - Education in Acoustics**
EL - Electroacoustics and Audio Engineering	**EL1 - Electroacoustics and Audio Engineering**
EN - Environmental Acoustics & Community Noise	**EN1 - Noise Assessment and Control.**
	EN2 - Noise Mapping
	EN3 - Road Traffic Noise Modeling and Noise Barrier
	EN4 - Smart City Sound Monitoring
	EN5 - Wind Farm Noise
MU - Musical Acoustics	**MU1 - Music Perception**
	MU2 - String Instruments
	MU3 - Numerical Computation in Musical Acoustics
	MU4 - Wind Instruments
	MU5 - General Musical Acoustics
NS - Noise: Sources and Control	**NS1 - Aircraft Noise - Aeroacoustics**
	NS2 - Hearing Protectors
	NS3 - Launch Vehicle Acoustics
	NS4 - Materials for Noise Control
	NS5 - Sustainable Materials for Sound Absorption and insulation
	NS6 - Noise: Sources and Control (others)

ICA 2016 CONGRESS TOPICS	SUBTOPICS (STRUCTURED SESSIONS)
NT - Numerical Techniques	**NT1 - Boundary Element and Meshless Methods on Acoustics and Vibrations**
	NT2 - Numerical Techniques (others)
PA - Physical Acoustics	**PA1 - Phononic crystals and acoustic metamaterials**
	PA2 - Sonochemistry and Sonoprocessing
	PA3 - Ultrasound
PP - Psychological and Physiological Acoustics	**PP1 - Free-Field Virtual Psychoacoustics and Hearing Impairment**
	PP2 - Product Sound Quality and Multimodal Interaction
	PP3 - Psychological and Physiological Acoustics (others)
SP - Signal Processing in Acoustics	**SP1 - Acoustic Array Systems**
	SP2 - Acoustic Array System: Near-field Acoustic Holography and Vibro-Acoustic Field Reconstruction
	SP3 - Model-Based Optimization/Estimation and Analysis
	SP4 - Signal Processing in Acoustics (others)
SS - Soundscape	**SS2 - Soundscape and Holistic Analysis**
	SS3 - Soundscape, Psychoacoustics and Urban Environment
	SS4 - Soundscape, Quality of Life , and Health
	SS5 - Spatial Sound Recordings in Preserved Habitats
SI - Acoustical Measurements and Instrumentation	**SI1 - Sound Intensity and Inverse Methods in Acoustics**
	SI2 - Acoustical Measurements and Instrumentation
SC - Speech Communication	**SC1 - Speech Communication**
SV - Structural Acoustics and Vibration	**SV1 - Structural Health Monitoring and Sensor Networks**
	SV2 - Structural Acoustics and Vibration (others)
UW - Underwater Acoustics	**UW1 - Underwater Acoustics**
VA - Virtual Acoustics	**VA1- Virtual Acoustics**

FIA 2016 Congress TOPICS	**AA - Acústica Arquitectónica - Acústica de Salas (Architectural Acoustics-Room and Building Acoustics)**
	EN - Acústica Ambiental y Ruido Comunitario (Environmental Acoustics & Community Noise)
	MU - Acústica Musical (Musical Acoustics)
	NS - Ruido: Fuentes y su Control (Noise: Sources and Control)
	PP - Acústica Psicológica y Fisiológica (Psychological and Physiological Acoustics)
	SP - Procesamiento de Señal en Acústica (Signal Processing in Acoustics)
	SS - Paisaje sonoro (Soundscape)

PROGRAM HIGHLIGHTS

Plenary Speakers



Michael Vorländer

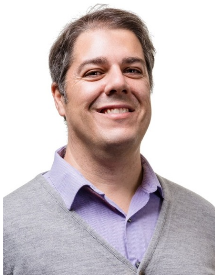
From acoustic simulation to virtual auditory displays

Simulation and auralization techniques are used in engineering, architecture, sound design and in applications in hearing research. The components of this technique are acoustic simulations and signal processing tools and the data interfaces in between, for which well-established solutions exist. The main bottlenecks are lack of data of 3D characterization of sound sources and material parameters, and interfaces to spatial audio technology. These problems are subject to research. Whether the virtual environment is considered sufficiently accurate or not, depends on many perceptual factors, and on the pre-conditioning and the degree of immersion of the user in the virtual environment. In this presentation the processing steps for creation of Virtual Acoustic Environments are briefly presented, and the achievable degree of realism discussed in examples including room acoustics, archeological acoustics, transportation noise, and hearing research.

Chen-Fen Huang

On the perspective of underwater acoustic tomography for probing ocean currents in shallow-water environments

Oceanographic processes in coastal regions including wind driven flows, tidal currents, river outflows, internal waves, eddies, western boundary currents, etc. are highly variable in time and space. Conventional oceanographic measurements (e.g., acoustic Doppler current profiler) cannot provide a synoptic image of those dynamic processes, especially for short time and space scales. Ocean Acoustic Tomography (OAT) uses time-of-flight measurements from different angles across the water. OAT is an effective method for mapping the spatial distribution of current and temperature fields. This talk will focus on the OAT applications to probe the current field in shallow water environments and present recent experimental results. Included are 1) the application of the middle-range (~50 km) OAT technique to study the spatial and temporal variations of the sub-branch of the Kuroshio off the east coast of Taiwan, 2) exploiting the communication signals of distributed networked underwater sensors for ocean current mapping, and 3) integrating moving vehicles to enhance OAT results.



Frank Russo

Understanding music perception from the perspective of oscillation and resonance

Over the last decade my lab has investigated psychoacoustic properties of pitch, timbre, and rhythm as perceived by the ear (auditory) as well as the skin (vibrotactile). Mechanoreceptors in the skin are structurally similar to those in the ear and exhibit frequency tuning enabling coarse pitch perception. Although the skin is equipped with only a few broadly tuned frequency channels and without a “place code”, this appears to be enough to enable discrimination between complex vibrotactile waveforms that have been matched for fundamental frequency and subjective magnitude (i.e., vibrotactile timbre perception). The skin is also quite capable of giving rise to the perception of rhythm, however this capacity proves challenging with complex rhythms. Neuro-electric measures allow us to examine resonance to different levels of oscillatory structure. Auditory neurons in the brainstem are capable of phase locking with tone frequencies in music. The fidelity of this type of neural resonance is better in individuals with music training and worse in individuals with hearing impairment. Neurons in auditory and motor cortices have been found to phase lock to the dominant beat frequency in music (i.e., the pulse). This form of neural resonance continues even after the music has stopped, and much like the brainstem response to tone frequencies, its fidelity tends to be better in individuals with music training. The picture that emerges from this body of work is that perception of music is underpinned by neural resonance to different levels of oscillatory structure present in auditory and vibrotactile waveforms. Long-term active engagement with music supports the fidelity of neural resonance.

Barbara Shinn-Cunningham

How the brain makes sense of complex auditory scenes

Everyday listening involves a complex interplay between the ear, which transduces sound energy into neural responses, and the brain, which makes sense of these inputs. Historically, research on the ear tended to ignore the fact that what we can perceive in sound depends on what task the brain is engaged by, while research on cortical processing of sound ignored the complexity and sophistication of how the ear works. In this talk, I will explore how everyday perceptual abilities depend jointly on how the ear encodes information (and individual differences in the fidelity with which it does so) and how attention and other state dependent variables change the information we perceive.

Samir Gerges

Hearing Protectors: State of the Art and Emerging Technologies of Comfort and Uncertainty in Measurements

In many industrial and military situations it is not practical or economical to reduce ambient noise to levels that present neither a hazard to hearing nor annoyance. In these situations, personal hearing protection devices are capable of reducing the noise by up to around 35 dB. Although the use of a hearing protector is recommended as a temporary solution until action is taken to control the noise, in practice, it ends up as a permanent solution in most cases. Therefore, hearing protectors must be both efficient in terms of noise attenuation and comfortable to wear. Comfort in this case is related to the agreement of the user to wear the hearing protector consistently and correctly at all times. The purpose of this paper is to review the stat of art for the need to develop methods to quantify comfort and noise leakage, also to quantify the uncertainty in evaluating hearing protector noise attenuation.

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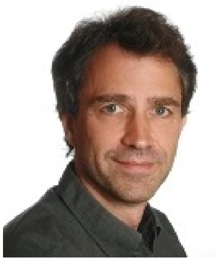
ICA 2016
ABSTRACTS

Monday, 5 September 2016

Monday morning, 5 September 2016 — **Juan Pablo II Auditorium**
Opening Ceremony
11:00 - 12:00

Monday midday, 5 September 2016 — **Juan Pablo II Auditorium**
12:00 - 13:00
Chair: Julio Cordiolli
Plenary Lecture



Michael Vorländer

[Paper ICA2016-481]

From acoustic simulation to virtual auditory displays

Michael Vorländer
Institute of Technical Acoustics, RWTH Aachen University, mvo@akustik.rwth-aachen.de

Abstract
Simulation and auralization techniques are used in engineering, architecture, sound design and in applications in hearing research. The components of this technique are acoustic simulations and signal processing tools and the data interfaces in between, for which well-established solutions exist. The main bottlenecks are lack of data of 3D characterization of sound sources and material parameters, and interfaces to spatial audio technology. These problems are subject to research. Whether the virtual environment is considered sufficiently accurate or not, depends on many perceptual factors, and on the pre-conditioning and the degree of immersion of the user in the virtual environment. In this presentation the processing steps for creation of Virtual Acoustic Environments are briefly presented, and the achievable degree of realism discussed in examples including room acoustics, archeological acoustics, transportation noise, and hearing research.

Noise Mapping:
Paper ICA2016-628

Reducing geometry or detailing it? Comparison between measured and modeled microscale urban spaces

Rafaella Estevão da Rocha[a], Stelamaris Rolla Bertoli[b], Alexandre Virginelli Maiorino[c]
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Abstract
One of the requirements for noise mapping elaboration is to reduce geometry due to its large-scale coverage and to its simplified algorithms methods of calculation. However, this simplification can neglect some important acoustic information that can influence the outdoor sound propagation. This research aims to compare the acoustics results between a reduced geometry model and a detailed one, verifying which scenario better represent the reality of the urban space. In order to do that, Room Acoustics software Odeon v13 was used, since traditional macroscale noise mapping software does not allow the production of accurate detailed geometry. Thus architectonic details were considered in the detailed model and scattering coefficient was gradually changed in the reduced geometry model. Simulation results were compared with *in situ* measurements using the Impulse Response technique. Analyzed parameters were T30, EDT and SPL. Energy-Time Curves were also compared. Results show that the change of scattering coefficients in the simplified model does not achieve the same good agreement with measurements as the detailed model. The addition of the details increases accuracy; therefore it can be assumed that the presence of the details can enhance the representation of the actual urban space.

Noise Mapping:
Paper ICA2016-401

Noise propagation software comparison: A case of study between SoundPLAN and Code_TYMPAN

Esteban Zanardi[a], Jorge Carrasco Henríquez[b], Jorge Torres[c]
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Abstract
In this research paper a comparison between the results obtained with SoundPLAN V7.3 and Code_TYMPAN V3.9 software, both based on calculation method specified in ISO 9613 norm, is made. Taking into account the differences between commercial noise propagation software and open source software, a calculation procedure is set to accomplish a fair comparison. A case study of industrial noise is presented with two different behaviour stages and a computer model for each one is developed. The main variation is the replacement of the old silencer with a new specially designed one. Noise measurements were carried out to compare results and to give initial values of sound power level from the industrial sources in the two stages. Conclusions are made according to the measured and calculated values.

Noise Mapping:

[Paper ICA2016-517]

Time and cost saving software techniques and their application in large-scale noise mapping projects

Antonio Notario

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Abstract

Noise mapping is one of the most important tools in the global fight against noise. The implementation of the Environmental Directive 2002/49/EC had been the main trigger in the development of techniques and software aiming to handle Noise Mapping projects in large areas or even complete cities. The implementation of an effective Policy against noise comprises several aspects from the introduction of a limiting value system, selection of the calculation methods, project management, up to the availability of results to the public in a clear, comprehensive and accessible way. On the other hand the time needed for and the quality of noise maps and action plans depend a lot on the software tools applied and therefore any modification of data which needs manpower can raise the time and cost enormously. A technique based on scripting is presented for automating time consuming tasks such as correcting models after the import of data from various third party formats, iterating over different configurations of calculations, summing up source-specific noise maps including level corrections or even exporting maps to an Online Noise Map Interface to make results accessible to the public.

Noise Mapping:

[Paper ICA2016-458]

ISO 1996 measurement procedure and the uncertainty associated in strategic noise maps

David Montes González[(a)], Juan Miguel Barrigón Morillas[(a)], Guillermo Rey Gozalo[(b)], Pedro Atanasio Moraga[(a)], Rosendo Vílchez-Gómez[(a)], Juan Antonio Méndez Sierra[(a)], Rubén Maderuelo Sanz[(c)]

[(a)] Departamento de Física Aplicada, Universidad de Extremadura, Cáceres, Spain, barrigon@unex.es

[(b)] Facultad de Ciencias de la Salud, Universidad Autónoma de Chile, Talca, Chile

[(c)] Departamento de Tecnologías y Construcción Sostenible, INTROMAC, Cáceres, Spain

Abstract

Strategic noise maps are an essential tool for the evaluation of the exposure of the population to noise pollution and the elaboration of Action Plans. In this regard, since *in situ* measures are required for the elaboration or the calibration and validation of noise maps, the Noise European Directive considers the standard ISO 1996 as a reference. On the one hand, this standard es-tablishes in its normative part some corrections as a function of the distance between the mi-crophone and the rear reflective surface. On the other hand, it contains an Annex B (informa-tive) in which certain conditions are established for each case in order that the values obtained by *in situ* measurements are approximate to these corrections. This paper show a review of the scientific literature about this topic, in which an analysis of published results and a reflection about the accuracy of the strategic noise maps carried out under the European Noise Directive are made.

Noise Mapping:

[Paper ICA2016-739]

Acoustic barrier for outdoor music event: The "roda de samba" in Rio de Janeiro

Lygia Niemeyer[a], Marina Cortês[b], Nayara Gevú[c]
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[c] Federal University of Rio de Janeiro - UFRJ, Postgraduate Program in Architecture - PROARQ, Brazil, nayaragevu@gmail.com

Abstract

Samba - Brazilian rhythm par excellence - brings in its origin a mix of rhythms and traditions that dates back the history of the country. To the drumming, African cultural heritage, were being gradually incorporated elements of other musical genres. The so-called "samba" consists of a group of singers accompanied by a suit of percussionists (bass drum, snare drum, and tambourine) guitar and ukulele. In Rio de Janeiro, due to climatic and geographical characteristics of the city, the *rodas de samba* are often held outdoors in clubs, bars or even on the street. The issue discussed in this paper is inserted in this context: the nuisance caused by a traditional *samba* wheel in residential surroundings. Created 10 years ago, the event that takes place on Mondays afternoon in a traditional Rio club courtyard was initially attended only by musicians and other professionals working on the weekend. In order to improve the life quality of the community and, at the same time, preserve one of the main symbolic elements of belonging and sound landmark. The simulations were performed with the SoundPLAN program in three different scenarios: current situation without event, current event situation and future situation (with barrier) with event. It was verified that in addition to music the audience voices, singing or talking, significantly contribute to the increase of level of sound pressure. The barrier performance was quite satisfactory, reducing the noise emitted into the environment without prejudice to the public on the courtyard of the club.

Monday afternoon, 5 September 2016 **Juan Pablo II Auditorium**
16:30 - 18:30
Environmental Acoustics & Community Noise
EN1 - Noise Assessment and Control

INVITED

Noise Assessment and Control:

[Paper ICA2016-723]

Low frequency noise and disturbance assessment methods: a brief literature overview and a new proposal

Marco Caniato[a], Federica Bettarello[b], Fausti Patrizio[c], Lucia Marsich[a], Alessio Ferluga[a], Chiara Schmid[a]
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Abstract

Several studies have presented the effects of environmental noise on communities, focusing the attention on the sleeping time events. The noise introduced into a dwelling is mostly evaluated using the A-weighted sound pressure level (LAeq) as the only parameter to determine the perceived disturbance. Nevertheless, if noise is produced by activities or sources characterised by a low frequency contribution, the measurement of LAeq underestimates the real disturbance, in particular during sleeping time. The aim of this contribution is to analyse the low frequency disturbance phenomenon into technical and scientific literature and to investigate if any possible objective method is present in order to assess noise disturbance inside dwellings.

Noise Assessment and Control:

Paper ICA2016-410

Analysis of environmental normative assessments of noise immision: Global values v/s spectrally detailed values

Jorge Carrasco Henríquez[a], Esteban Zanardi[b], Jorge Torres[c]
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[b] DECIBEL SUDAMERICANA S.A., Argentina, estebanzanardi@decibel.com.ar
[c] DECIBEL Chile Ingeniería Acústica Ltda., Chile, jorge.torres@decibel.cl

Abstract

This paper aims to realize a comparative exercise between different methodologies for the evaluation of environmental noise immission. Environmental noise evaluated according to procedure specified in ISO 1996, currently used in Europe, indicates that measurements must be realized by third octave, as well as establishing a method to analyse low frequencies components, tonal components and impulse components. Meanwhile, most south American normative tend to evaluate the environmental noise immission based exclusively on global values measured and correct them depending on the background noise. In certain cases corrections are also applied to inside measures taking into account if measurements were realized with open or closed windows or vains. Three different case analysis applying the two different normative evaluations were carried out, one of them based on measures in accordance with ISO 1996 and penalizations specified in O.M.A. (Ordenança Mediambiental de Barcelona) and the other one in accordance to procedure and penalizations described in D.S. N°38/11 MMA of Chile. Additionally, the recently updated Argentinean IRAM 4062, which includes certain qualification similar to ISO 1996, is considered. Analysis of the exigencies of environmental noise between the norms is made and conclusions are taken regarding the noise assessment for exposed communities.

Noise Assessment and Control:

Paper ICA2016-378

Effects of nocturnal air and rail traffic noise on sleep

Uwe Müller[a], Eva-Maria Elmenhorst [a], Franco Mendolia[a], Mathias Basner[b], Sarah McGuire[b], Daniel Aeschbach[a]
[a] Division of Flight Physiology, Institute of Aerospace Medicine, German Aerospace Center DLR, Cologne, Germany, Uwe.Mueller@dlr.de
[b] Division of Sleep and Chronobiology, Department of Psychiatry, Perelman School of Medicine, University of Pennsylvania, Philadelphia, PA, USA, Basner@mail.med.upenn.edu

Abstract

Undisturbed and sufficiently long sleep is a prerequisite for a healthy life as well as for the prevention of fatigue-induced accidents. Especially the increasing air and freight rail traffic is more and more shifted to shoulder and night-time hours due to missing capacity and infrastructure during daytime. Thus, the sleep of residents near airports or railway tracks is increasingly affected by traffic noise. Only very few main airports, such as Frankfurt (Germany), implemented a night flight ban in order to countervail this trend. Since 1999 the Institute of Aerospace Medicine of the German Aerospace Center (DLR) has investigated these night time noise effects in several field studies in which the sound pressure levels L_{AS} and L_{AF} and sound files were continuously measured with class one sound level meters at the sleeper's ear. Sleep structure was recorded with polysomnography (simultaneous measurement of brain waves, eye movements, and muscle tone), the gold-standard to quantify sleep objectively.The results on sleep quality and additional awakening reactions due to traffic noise from former studies performed at Cologne/Bonn airport (high night time traffic) and a busy railway track in the Rhine valley (high night time freight traffic) are compared with the results of the recently completed NORAH (Noise-Related Annoyance, Cognition, and Health) study at Frankfurt airport. In the latter study data were collected both before as well as after the implementation of a ban of night flights between 11 p.m. and 5 a.m.. Sound exposure distributions, average sound levels and sound level rise time distributions at the sleepers' ear are presented for all three studies.

Noise Assessment and Control:

[Paper ICA2016-218]

Tests of influence of high frequency noise on human psycho-physical efficiency

Bozena Smagowska

Central Institute For Labour Protection - National Research Institute, Poland, bosma@ciop.pl

Abstract

The Noise with components of high audible frequency (10-16 kHz) and low ultrasonic frequency (20-40 kHz) is defined in Poland as ultrasonic noise. This noise is a harmful factor which causes annoyance and dangerous effects on the human body, particularly in the working environment. The paper presents the method and results of laboratory tests carried out to determine the influence of high frequency noise on psycho-physical efficiency of workers. The method was based on the change in mental capacity of subjects exposed to ultrasonic noise in such function as reflex, observation skills, attention and work output, subjective estimation of mood and tiredness of the subjects. These functions were assessed with indicators from psychological tests and on the basis of questionnaires. The tests were conducted in three variants of acoustics conditions (without noise, with a tonal noise, with a broadband noise). The results showed that participation of subjects in different conditions of experiments had a considerable influence on subjective assessment of mood and tiredness. The tests that were carried out made it possible to determine preliminary proposal for annoyance criterion of ultrasonic noise for third octave bands with center frequency from 10 to 40 kHz for activities which require the focus of attention.

Noise Assessment and Control:

[Paper ICA2016-216]

Using a smartphone application to perform a fast qualitative noise exposure evaluation in a mine site

Luis Corral[a], Pierre Aumond[b]

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[b] Compañía Electroacústica Sudamericana LTDA, Chile, pierre.aumond@gmail.com

Abstract

The Sensorineural Hearing Loss presents a high prevalence in the population, being one of the most important professional diseases. The number of people in the world affected by this pathology is estimated near 360 million. In Chile, in 2011, the Protocol of Occupational Noise Exposure (PREXOR) was released, which establish ambient and health vigilance programs for the workers exposed to occupational noise. In this legal framework, in 2015 the Exempt Resolution 859 was released which involve the implementation of qualitative noise exposure evaluations. In this work, an Android platform mobile application is presented, which aims to ease and develop the implementation of this protocol. The data collected by the application are sent to a web interface where PREXOR's compatible qualitative files can be downloaded. The results from qualitative noise exposure evaluation in a mine site are presented.

Noise Assessment and Control:
Paper ICA2016-157

NORAH (Noise Related Annoyance, Cognition, and Health): Questions, designs, and main results

Rainer Guski[a], Maria Klatte[b], Ulrich Moehler[c], Uwe Müller[d], Anja zur Nieden[e], Dirk Schreckenberg[f]
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Abstract
The German multidisciplinary research project NORAH (Noise Related Annoyance, Cognition and Health) was aimed at providing a broad and scientifically reliable description of the effects of air, road and rail traffic noise on the health and life quality of residents in the vicinity of airports. Ten scientific institutes participated and performed surveys, secondary health data analyses, sleep quality registrations, blood pressure registrations, and special tests on children at school. Main results: 1. At all four airports studied, the percentage of persons highly annoyed by air traffic noise at comparable noise levels was larger than would be expected from the so-called "EU standard curves" [1]. 2. With respect to cardiovascular health risks, the effects of rail and road traffic noise on heart failure, myocardial infarction, and stroke were more clearly seen as compared to the effects of aviation noise. 3. There was no statistically significant increase of self-registered blood pressure values with increasing LpAeq for the evening and night-time for transportation noise.4. Night-time sleep of residents showed a diminished number of aircraft associated awakenings with the introduction of the night curfew at Frankfurt Airport for a group being in bed during 22:00-22:30 hrs until 06:00-06:30 hrs. The probability of awakening due to a single aircraft event, however, did not change before and after the night curfew. 5. Multilevel analyses revealed a significant linear association between aircraft noise levels at school and decreasing reading performance in second graders. A one month delay in reading was observed for an increase in noise levels by 10 dB LpAeq.

Monday afternoon, 5 September 2016 **Dr. Valsecchi Auditorium**
14:30 - 16:10
Biomedical Acoustics
BA1 - Biomedical Acoustics

INVITED
Biomedical Acoustics:
Paper ICA2016-1135

Biomedical acoustic imaging sensors for U-health care applications

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Abstract
Day by day, there is an advancement in sensor technology and newer and newer sensors are being developed, for getting new industrial and biomedical applications. However, practical clinical aspects of such sensors on the patients are still required to be studied in detail. Here, advanced acoustic imaging sensors are presented for better health care, in a ubiquitous manner. Design and development of novel electro-acoustic, MEMS based piezo-resistive, piezoelectric and piezo-composite types of nano-imaging systems, along with WSN and U-technology, are presented, for the health care of old age patients, particularly for those living in isolated environment. Main emphasis is placed on the nano-cancer technology, POCT devices and bone-based diagnostic acoustic imaging systems for the quick diagnosis of abnormalities/diseases of critically ill patients, for better U-healthcare.

Biomedical Acoustics:

[Paper ICA2016-148]

A study on the feasibility of MEMS piezoelectric accelerometer coupled to the middle ear as sensor for totally implantable hearing devices

Andre Gesing[a], Diego Calero[a], Bernardo Murta[a], Stephan Paul[a], Julio Cordioli[a]
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Abstract

The presence of an external element is still a major limitation of current hearing devices such as hearing aids and cochlear implants. The main problems associated with the external element are discomfort, inconvenience and social stigma, which can be overcome by totally implantable hearing devices. A fundamental requirement of such systems is a totally implantable sensor. In this sense, an accelerometer coupled to the ossicular chain of the middle ear may be used as an alternative sensor to the traditional external microphone. Although micro machined accelerometers are used in a variety of applications, there are no commercially available accelerometers that fulfill the requirements. This paper presents a review on implantable sensors for hearing devices, and requirements for this transducers are summarized. The Finite Element Method (FEM) is used to verify the feasibility of an implantable piezoelectric accelerometer. Design was made considering limitations and characteristics of microelectromechanical systems (MEMS) fabrication techniques. Different designs for the accelerometer are considered, and the results for the sensor response at different points of the ossicular chain are presented and analyzed in view of the defined requirements.

Biomedical Acoustics:

[Paper ICA2016-563]

Photoacoustic imaging of vasculature with parabolic array transducer

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Abstract

Ideal photoacoustic (PA) signal is a spherical wave generated from a point source. However, realistic PA signal is highly dependent on the structure of the target. For example, when the laser is irradiated to the tubular structure, most of the PA signal is generated from the surface of laser irradiated side. Multi-angle irradiation of the laser or multi-angle detection of the PA signal is required for compensation of the PA feature. In the present study, PA imaging system with parabolic array transducer is developed for receiving the PA signal from multiple angle. Parabolic array transducer was consisted of 256-ch 1-3-composite elements with the central frequency of 10 MHz. The diameter was 42.4 mm and the opening angle was 90 . A hole of 10.4 mm diameter was made in the central part of the transducer for transmission of the laser. PA signal was received by ultrasound data acquisition system with 256 transmit channels and 256 receive channels. The laser with the wavelength of 532 nm, the power of 1.06 mJ/cm2 and the repetition rate of 10 Hz was equipped for generation of PA signal. First, the resolution of the system was tested by the observation of a hair phantom with the diameter of 80 mm immersed in water. The full width half maximum of the ultrasound imaging was 130 m and that of PA imaging was 70 m. Second, more realistic model was made with the vasculature at the surface of cod roe immersed in lipid emulsion. The vasculature was clearly shown by the PA imaging system. PA imaging with parabolic array transducer successfully visualized the vasculature of cod roe. The PA system may visualize the capillary in human skin although the central frequency was 10 MHz.

Biomedical Acoustics:

[Paper ICA2016-284]

The variable cochlear hydro-mechanical inertance

Santos Tieso[a], Lucas Fantini[a], Francisco Messina[a], Nahuel Cacavelos[a], Gilda Farelli[a], Leonardo Zavala[a], Maria Tieso[a], Sebastian Iezzi[a], Nicolás Casco Richiede

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Abstract

In this paper, the way in which the cochlear changes its inertance when stimulated by different frequencies is studied and explained. This phenomenon allows the human's ear to have an eleven octave frequency range. The cochlear hydro-mechanical inertance is defined as the density of the liquid contained inside the cochlea, set in motion with a particular geometry. This geometry is modified by histo-anatomic structures in response to the different stimulus frequency. The study of the fluid dynamics inside the cochlea will allow a deeper understanding of the way the mammal's ears work as well as providing new ways to treat diseases or injuries.

Biomedical Acoustics:

[Paper ICA2016-405]

On the relation between pressure applied to the chest piece of a stethoscope and parameters of the transmitted bioacoustic signals

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Abstract

The force with which the chest piece of a stethoscope is pressed against the body of a patient during an auscultation examination introduces the initial stress and deformation to the diaphragm and the underlying tissues, thus altering the acoustic parameters of the sound transmission path. If the examination is performed by an experienced physician, he will intuitively adjust the amount of the force in order to achieve the optimal quality of the heard sound. However, in case of becoming increasingly popular auto-diagnosis and telemedicine auscultation devices with no instant feedback mechanisms which could perform such an adjustment procedure, the question arises regarding the influence of the possible force mismatch on the parameters of the recorded signal. The present study describes the results of the experimental investigations on the relation between pressure applied to the chestpiece of a stethoscope and parameters of the transmitted bioacoustic signals. The experiments were carried out using acoustic and electronic stethoscopes connected to the developed and constructed force measurement system, which allowed to maintain a given value of the applied pressure during auscultation examinations. The signals were recorded during examinations of different volunteers, at various auscultation sites. The obtained results reveal strong individual and auscultation-site variability. It is concluded that the underlying tissue deformation is the primary factor that alters the parameters of the recorded signals. It is shown, that in certain cases applying too light or too firm pressure to the chest piece may result in significant decrease of specific frequency components. Possibilities of developing universal force control algorithms without feedback mechanisms are discussed.

INVITED

Biomedical Acoustics:

Paper ICA2016-630

Advances in cortical bone assessment using ultrasonic resonances and guided waves

Jean-Gabriel Minonzio[a], Quentin Vallet, Nicolas Bochud, Quentin Grimal, Pascal Laugier

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Abstract

Assessment of bone mechanical properties is an important clinical issue. Recently, progresses achieved in quantitative ultrasound have stimulated a renewed interest for basic and clinical bone studies. Resonant ultrasonic spectroscopy has been adapted for the full characterization of the anisotropic stiffness of small specimens of highly damping materials such as cortical bone. In RUS, the elastic properties are estimated by solving an inverse problem by fitting model predicted frequencies to the measured resonant frequencies. We have introduced a Bayesian approach in which an a priori knowledge of the exact pairing between measured and predicted resonant frequencies is not necessary, the optimal pairing being determined in the course of the optimization process. RUS is prone to provide answers to questions that remain open regarding the determinants of cortical bone elastic properties. On the other hand, guided waves measured in axial transmission, have been proposed for the *in vivo* investigation of appendicular bones (tibia, radius) which are relatively accessible to measurements. We have developed a specific data processing and an inverse problem solving scheme using genetic algorithms to overcome the challenges caused by surrounding soft tissues and by the complex structure of the cortical waveguide. Our procedure allows estimating cortical bone biomarkers such as cortical thickness, stiffness and porosity.

Biomedical Acoustics:

Paper ICA2016-642

Observing cytoskeletal changes in cancer cells using high-frequency (10-100 MHz) ultrasonic spectroscopy

Caitlin Carter[a], Ashley Behan[a], Dolly Sanjinez[a], Amy Lafond[a], Timothy Doyle[a]

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Abstract

The cytoskeleton is pivotal to the biomechanical properties of cells. It therefore plays a crucial role in the behavior and progression of cancer. Cytoskeletal changes can enable cancer cells to become more mobile, thereby facilitating their infiltration into tissue or metastasis to other parts of the body. Cytoskeletal anomalies can also be associated with specific molecular subtypes of a cancer. For example, the more aggressive subtypes of breast cancer, such as basal-like and Her2+, have mutations that alter the protein regulation of the cytoskeleton. These subtypes may, therefore, be detectable via their effect on the cytoskeleton and cell biomechanics. The objective of this work was to determine if high-frequency (10-100 MHz) ultrasonic spectroscopy can detect chemically induced changes in the cytoskeleton of cancer cells. Cell cultures of a human pancreatic carcinoma cell line (panc-1) were grown in monolayers and then treated with sphingosylphosphorylcholine (SPC), a bioactive lipid that rearranges the keratin components of the cytoskeleton. Continuous pulse-echo measurements of the cultures were taken over a period of one hour. Computer simulations were performed to verify the results. The simulations modeled the ultrasonic spectra based on the internal structure of the cells using a multipole expansion method. The experimental spectra showed changes that were consistent with the simulated spectra and the optically observed changes in the keratin network. The results of this research demonstrate that high-frequency ultrasonic spectra are sensitive to cytoskeletal changes in cancer cells induced by SPC.

Biomedical Acoustics:
Paper ICA2016-643

Sensitivity of high-frequency ultrasound to breast cancer lobular carcinomas: results from phantom and surgical specimen studies

Nicole Cowan[a], Zachary Coffman[a], Robyn Omer[a], Timothy Doyle[a]
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Abstract
A majority of women with early stage breast cancer select breast conservation surgery (BCS) over mastectomy. A key issue with BCS, however, is the high percentage of patients (30-60%) who require additional surgery to remove residual cancer that was not identified during the initial operation. This is especially true for patients with lobular carcinomas since they are difficult to detect. At Utah Valley University, a high-frequency (HF) ultrasonic method has shown promise as a rapid, intraoperative method for detecting residual breast cancer in surgical margins. The objective of this project was to determine the sensitivity of HF ultrasound to lobular carcinoma using histology mimicking phantoms and surgical margins. Phantoms were created from distilled water, agarose powder, 10X TBE stock solution, and polyethylene microspheres (98-μm dia.) and fibers (35-μm dia.) to simulate breast tissue histology. Three experiments were conducted with specimens containing only microspheres (E1), only fibers (E2), and a mixture of both (E3) to more accurately model breast tissue histology. Microsphere and fiber weight percents were varied for each specimen. Pitch-catch measurements were acquired using 50-MHz transducers, a HF ultrasound system, and glycerol as the coupling agent. Attenuation showed definite trends for E1 and E3, but no trend for E2. Peak density showed no trend for any of the experiments. HF ultrasonic tests on margin specimens from two studies performed at the Huntsman Cancer Institute (90 patients total) showed that both attenuation and peak density were most sensitive to lobular carcinomas. Both phantom and surgical margin results indicate that HF ultrasound shows a higher sensitivity to lobular carcinoma histologies as compared to ductal carcinoma histologies.

Biomedical Acoustics:
Paper ICA2016-646

Histology mimicking phantoms for the high-frequency ultrasound of breast cancer surgical margins: Comparison between gelatin and agarose media

Zachary Coffman[a], Nicole Cowan[a], Robyn Omer[a], Timothy Doyle[a]
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Abstract
At Utah Valley University, a high-frequency (HF) ultrasonic method has shown promise as a rapid, intraoperative method for detecting residual breast cancer in margins resulting from breast conservation surgery (BCS). Due to the difficulty of obtaining and routinely testing human tissue samples in a laboratory setting, soft tissue phantoms have been developed with inclusions to simulate the microstructures and histology of breast tissue pathologies. The objective of this study was to determine the optimal phantom medium to maximize the fidelity of the phantoms to actual breast tissue and its histology. Gelatin based phantoms were made from a mixture of distilled water, Knox gelatin, and Metamucil fiber while agarose based phantoms were created from distilled water, agarose powder, and TBE stock solution. In both phantom mediums polyethylene microspheres were embedded in layers in order to simulate breast tissue microstructure. Microsphere diameter varied by phantom while a constant volume percent was maintained. Pitch-catch measurements were acquired using 50-MHz transducers, a HF pulser-receiver, a 1-GHz digital oscilloscope, and glycerol as the coupling agent. Results from the gelatin phantoms showed a decrease in both peak density and attenuation values with increasing microsphere diameter and overall decreasing heterogeneity of the phantoms. The agarose phantoms showed the same results with comparable standard deviations. These results not only indicate that both gelatin and agarose based phantoms can be effectively used to accurately simulate breast tissue microstructure and pathology, but that the optimal phantom medium may be based on personal preference or experimental needs.

Biomedical Acoustics:

[Paper ICA2016-647]

Evaluating margin status with high-frequency (20-80 MHz) analytical ultrasound during breast conservation surgery

Robyn Omer[(a)], Amy LaFond[(a)], Caitlin Carter[(a)], Leigh Neumayer[(b)], Rachel Factor[(c)], Timothy Doyle[(a)]

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Abstract

A majority of patients with early stage breast cancer elect breast conservation surgery (BCS) since it preserves unaffected breast tissue and, when followed by radiotherapy, provides survival rates equal to those of mastectomy. To ensure all of the cancer has been removed, excised tissue is sent to a pathology lab for analysis to determine if residual cancer is present in the margins (the boundary between resected and unresected tissue). This analysis can take up to several days. Unfortunately, 30-60% of BCS patients undergo additional surgeries to excise residual cancer not identified and removed during the initial surgery. The development of an intraoperative method to evaluate resected margins is, therefore, a crucial priority in breast cancer therapy. The objective of this work was to determine the sensitivity and specificity of high-frequency (20-80 MHz) analytical ultrasound for detecting residual cancer in margins by conducting a 17-patient pilot study and 73-patient validation study at the Huntsman Cancer Institute, Salt Lake City, Utah. Point measurements at 775 positions were collected from 383 resected margin specimens in through-transmission mode using 50-MHz, 6.35-mm diameter, single-element transducers. Attenuation and peak density (the number of peaks and valleys in a specified frequency band) were calculated from the ultrasonic waveforms and power spectra, respectively, and the two parameters were combined to perform a multivariate analysis. The pilot and validation studies showed sensitivity values of 87.5% and 87.0%, respectively, and specificity values of 82.9% and 67.2%, respectively. The results demonstrate that high-frequency ultrasound provides excellent sensitivity and good specificity for the rapid, intraoperative evaluation of BCS margins.

Biomedical Acoustics:

[Paper ICA2016-651]

Detecting breast cancer with high-frequency (20-80 MHz) ultrasound: A histological perspective

Timothy Doyle

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Abstract

The development of intraoperative methods to detect breast cancer in excised tissue specimens is a crucial priority in breast conservation surgery (BCS). A finding of cancer in tissue margins (the boundary between resected and unresected tissue) by pathology analysis after surgery indicates that some of the cancer was missed during BCS and the patient must return for additional surgery. Evaluating sentinel lymph node status during surgery is also critical since it determines the need for axillary lymph node dissection. High-frequency ultrasound has been found to display high sensitivity and specificity for malignant tissue in BCS margin specimens and lymph nodes. The method uses through transmission of narrow pulses centered at 50 MHz, yielding a broad spectral response of the tissue sample. The received pulses produce noise-free spectra that vary in shape with tissue pathology. BCS specimen data indicate that the peak density of the spectrum (the number of peaks and valleys in the 20-80 MHz band) correlates with breast tissue histology, but not with mammographic tissue density. Experiments with histology mimicking phantoms containing microspheres and fibers show that peak density is insensitive to fibrous microstructures but strongly correlates to microsphere size. Computer simulations of forward scattering from lobular carcinoma *in situ* models show that peak density increases with tumor progression, but also suggest that peak density arises from scattering resonances with cells. Thus, the sensitivity of peak density to cancerous tissue may be due to structural differences between malignant and normal cells. Whether it characterizes small-scale tissue structure or cell properties, peak density offers a promising window into the histology and detection of breast cancer.

INVITED

Challenges and Solutions in Acoustical Measurements and Design:

[Paper ICA2016-621]

Modeling and measurements of aircraft noise for single-family houses to meet local ordinance

Christopher Barnobi[(a)], Mike Greene[(b)], Adam Young[(c)], Arno Bommer[(d)], Robert Bruce[(e)], Isaac Harwell[(f)]

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Abstract

Residential noise from aircraft operations has received a lot of attention since the development of commercial air travel. This paper presents the results from modeling and measurements of aircraft noise for a single family housing development located near an airport. Modeling efforts compared two HUD standard methods for calculating indoor noise levels with a more detailed method. Modeling efforts also address different floor plans and constructions. Measurements of fly-over events were also conducted in model houses. The data was used to show conformance of the interior sound levels to a local ordinance. Further analysis of the data confirms/reveals other details about aircraft noise in single family houses. Additionally, a similar project included the use of an acoustic camera. The acoustic camera data can be used to look for weak spots in exterior wall/window construction during fly-over events.

INVITED

Challenges and Solutions in Acoustical Measurements and Design:

[Paper ICA2016-658]

Case studies of concave rooms for speech and music: measurement and design

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Abstract

Two case studies are presented for rooms with circular/oval shape. The newly constructed St. Dominic Chapel in Jackson, Mississippi (a clover leaf), and the newly renovated Poindexter Hall, built in 1905 and known originally as “The Temple of Music”, at the Mississippi University for Women in Columbus, Mississippi (an oval). Challenges and solutions to meet the acoustical requirements are for St. Dominic's chapel are presented with final room performance data. Poindexter Hall exhibited its major acoustic anomaly during demolition field testing with a double bass, which was later easily extracted from the sine sweep test data, but buried in the broadband data. Design solutions and measurements of the renovated Poindexter Hall are presented.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-50

The simplified method versus the detailed method of calculating flanking sound transmission through walls with linings

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Abstract
The simplified and detailed methods of calculating the apparent sound reduction index according to the standard, ISO 15712 have often been claimed to result in similar values for the weighted apparent sound reduction index. However, in extended studies on walls with linings at the National Research Council Canada, it has been found that the simplified method to calculate the flanking transmission through building elements with linings sometimes leads to misleading results. An alternative method for calculating the flanking sound transmission through walls with linings was proposed to ensure that the simplified method yields more conservative results than the detailed method. To achieve the best possible estimate of the sound insulation performance of buildings systems with linings, the detailed method should be used.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-121

Stravinsky Hall of the Moscow Musical Theatre “Helikon-Opera”: Acoustic challenges and achieved results

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Abstract
The musical theatre Helikon-Opera was founded in Moscow 26 years ago. Today the theatre is extremely popular not only in Russia, but even abroad. In 2007 a unique reconstruction of some historical buildings in the centre of Moscow started. Main challenge of this project was an adjustment of the courtyard for the opera hall equipped with the latest theatrical facilities. From acoustic point of view proposed concept had some difficulties. First of all the hall width is significantly over its length. Secondly it is the hall’s huge volume, which is about 7000 cubic meters. At the same time it is designed for 500 seats only. Moreover, historical view of the courtyard walls should be entirely saved. So acoustic design was highly constrained but some improvements of its acoustic properties were realized. In this paper we present detailed description of the hall design and its acoustic features. Proposed changes in hall design based on the simulations and their influence on hall acoustics are given as well. After completion of the reconstruction acoustic parameters of the hall, stage and orchestra pit were measured. The most interesting result is relatively long reverberation time and good speech intelligibility. We discuss characteristics of the hall and compare it with other opera houses. On November 2, 2015 new hall for opera performances named Stravinsky Hall was officially opened. During several months after opening different subjective evaluations from soloists, musicians, conductors and spectators were collected. They are cited in the paper as well.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-483

Speech Coherence Index: An intrinsically source independent method to estimate intelligibility

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Abstract
A method to estimate speech intelligibility has been developed that mimics the multi-resolution nature of human hearing and can be used with any non-repeating input signal. Previous intelligibility metrics

are defined and implemented assuming that background noise is measured separately or that the transfer function is measured with a specific signal, such as modulated noise, Maximum Length Sequence (MLS), or sine sweep.The new metric uses the coherence function, which is related to the complex-valued frequency response and can directly yield a signal-to-noise ratio (SNR) in the frequency domain, given a known input signal and a measured output signal. The coherence function is sensitive to any energy not present in the input, whether the extra energy comes from distortion, reverberation, or background noise. Significantly, coherence can be calculated in real time for any source or any noise level, even if the signal level changes with time. When used with different analysis lengths for different frequencies (multi-resolution), as is common practice in the audio industry, coherence can be used to replace the modulation transfer function in the Speech Transmission Index (STI) standard. The new metric is called Speech Coherence Index (SCI), and is compared to STI under simulated conditions with uncorrelated noise; with a single reflection; and with synthesized reverb, uncorrelated noise, and changes in direct level. The SCI value responds similarly to the STI, but is more consistent under certain conditions.

Monday afternoon, 5 September 2016 **Cardenal Pironio Auditorium**
16:30 - 18:30
Architectural Acoustics - Room and Building Acoustics
AA5 - Challenges and Solutions in Acoustical Measurements and Design

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-89

Analysing, quantifying and improving the sound transmission between two performance halls in a common building

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Abstract
An existing concert hall in Utrecht, a town in the Netherlands, has been integrated in a new building, creating a music centre with five performance halls for different types of music. In order to be able to use the various halls simultaneously the halls were designed according to the box-in-box principle, partly with light weight constructions, and completely separated from each other to achieve an outstanding sound reduction between them. For most situation this proved to be achieved, but between the halls for Pop music and Chamber music it failed by about 7 dBAin relation to heavy low frequency pop music. The authors were asked to analyse the problem, determine the dominant transmission paths and design solutions. By considering all possible transmission paths between these halls and quantifying the transmission by calculations and measurements, it was possible to identify a dominant path through columns supporting one of the halls. Reducing this transmission by applying resilient elements in the columns was predicted to give an improvement by 5 dB(A). In taking these measures many problems needed to be addressed, like the static load, the varying load by people in the hall and the vertical and horizontal dynamic stiffness of the elements. But it worked, measurements afterwards confirmed the predicted gain in sound reduction between the halls.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-337

Modified wave equation for modelling diffuse sound fields

Hugo Dujourdy[a,b], Baptiste Pialot[b], Thomas Toulemonde[a], Jean-Dominique Polack[b]
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Abstract
To the design of room, acousticians normally use besides their experience different tools such as computer modelling software. The most used is ray-tracing technique but it is not sufficient for taking into account the particular shape of long rectangular rooms. For example, in such rooms, most angles of reflection are large, with absorption coefficients that do not correspond to ISO random incidence

absorption values. This leads to inefficient models. Moreover, the same applies to the scattering process though furniture. Frequency bandwidth is another barrier for accurate acoustical prediction with ray-tracing technique. This paper presents a modified wave equation for sound energy density and sound intensity. By revisiting the relationships between those two quantities, we model the physical phenomena involved by sound propagation in a finite medium with obstacle as a modified wave equation. This linear second-order hyperbolic equation depends on few parameters such as absorption and diffusion coefficients. We propose an adjustment method of the model with in situ measurements to estimate the coefficients in an one-dimensional case.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-379

What is “proximity”, how do early reflections and reverberation affect it, and can it be studied with LOC and existing binaural data?

David Griesinger
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Abstract

Lokki has recently shown that there is no correlation between most current ISO 3382 hall measurements and preference, and that the most prominent perception preferred by his listeners is currently unnamed and unmeasurable. He chose to name the perception “Proximity” because he found it was related to the auditory sense of being close to the performers. This paper proposes that proximity arises from the phase alignment of the upper harmonics of speech and most musical instruments. We present data from other fields that shows that the loss of phase alignment due to early reflections or masking can greatly decrease the ability to separate signals from noise and other signals. We will then show how convolving some existing binaural data from Boston Symphony Hall with Lokki’s anechoic recordings can create a realistic binaural rendition of an instrumental ensemble, which can be used to test the effects of early reflections on proximity, localization, and loudness in these seats. We find that in all our seat positions attenuating the side wall reflection in Boston either improves the sound, or is (in very good seats) inaudible. These effects are predicted by the author’s binaural measure LOC.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-381

Playback of non-individual binaural recordings without head tracking, and its potential for archiving and analyzing concert hall acoustics

David Griesinger
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Abstract

Visually blind, rapid A/B comparisons are essential for studying auditory perceptions of all kinds, but are difficult to achieve in concert halls. Accurate binaural recording potentially allows instant A/B comparisons between halls and seats, and could verify that a laboratory simulation works as intended. But to be useful a standardized binaural recording needs to be accurately reproduced for a large variety of people. We have developed a computer-based loudness matching application that can quickly and non-invasively achieve individual headphone equalization with a variety of phones. When these phones are used to reproduce a free-field equalized microphone timbre is precisely preserved, and no head tracking is needed. The result is an uncanny impression of being exactly in a particular seat in a given hall. Halls and seats can be A/B compared with both live music and impulse responses from a virtual orchestra. These impulse responses can be manipulated to test the effects of different reflections, stage conditions, and reverberation times. By binaurally recording the sound in a particular seat a loudspeaker simulation of the same seat can be verified by taking the headphones on and off.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-843

Room acoustic experiments inside the Universidade Federal de Santa Maria Industrial College: A case study with low-cost instrumentation

Jean Carlo Bernardi[a], **Bruno G. Knebel**[b], **Bernardo H. Pereira Murta**[c], **William D'A. Fonseca**[d], **Paulo H. Mareze**[e], **Eric Brandão**[f]
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Abstract

When studying Room Acoustics, simulations simply are not enough to get the whole picture of the subject. It is important to notice that, measurements on site are also a vital part of the learning process, since it allows the estimation of relevant information concerning the rooms' acoustic functional efficiency. Later, this information can be used to evaluate acoustic parameters that will play an important role in the acoustic quality for the room. It is very important for students to get experience with rooms acoustic measurements. Nevertheless, it cannot always be carried out easily. Among several reasons, one is the high price of the professional acoustic equipment needed to properly do so. Facing this problem, this work aims to present a case study using lowcost equipment as an alternative method to grant more learning freedom for the students. The case study takes place at Universidade Federal de Santa Maria's Technical Industrial College Auditorium, Brazil, where the students performed several binaural measurements to evaluate the Auditorium's performance related to speech and music.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-463

Analysis of acoustic devices used in homestudio

Eduardo Silva[a], **Lúcia Oiticica**[b]
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Abstract

Homestudios are places installed in residential environment where is possible to produce music. The advance of technology contributed to the professionalization of these spaces, demanding larger efficiency mainly in acoustic performance in order to develop better phonographic materials in all steps of musical production. The objective of this work is to analyze acoustic devices installed in a chosen residential room for musical production activities: Helmholtz resonators for low frequency control, porous absorber for middle and high frequency controls, and acoustic diffusers to guarantee the homogeneous distribution of sound waves, promoting the adaptation of the chosen room to the ideal and technical recommendations for the control of internal acoustics phenomena in small rooms for musical activities. The methodology applied was prepared in these steps: documental search; architectural measurements; theoretical acoustic calculation in a spreadsheet; acoustic measurements of a residential room with a spectrum analyzer software based on Fast Fourrier Transform (FFT); proposal of construction of acoustic treatment devices and it's installation; financial viability analysis; and analysis of the influence of each of them in the space chosen for its application in arrangements and individually. The results showed that these acoustic control equipments attended to all of the necessary requisites for attenuation of the main prejudicial acoustic phenomena in the homestudio, although, it was found that the performance of the Helmholtz resonators were just guaranteed when applied together with other devices, mainly with diffractal diffusers, because of their capability to spread sound high pressure zones in the ambient.

Speech Communication:
Paper ICA2016-420

How long is a vocal tract? Comparison of acoustic impedance spectrometry with magnetic resonance imaging

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Abstract
Acoustic impedance spectrometry using the three-microphone, three-calibration technique has recently been applied to the vocal tract during phonation (Hanna *et al*., 2016. JASA, 139, 2924–2936). The qualitative and quantitative similarity of the impedance spectrum of the vocal tract with a simple cylindrical duct prompts the question: How well do geometric parameters derived from the measured impedance correspond to the vocal tract morphology? The main aim of this study is to compare, in one male subject (age 34, height 184 cm), the effective acoustic length of the vocal tract derived from impedance spectrometry with the anatomical length measured from a separate magnetic resonance imaging (MRI) scan. Three conditions were studied: 1) acoustic impedance measurements while the subject performed a neutral /ɜ/ vowel gesture with the lips sealed around the impedance measurement head, 2) MRI scan acquired while the subject performed the same gesture with a section of pipe between his lips of the same dimensions as the impedance head, and 3) MRI scan during closed-mouth nasal breathing. Even for the neutral vowel, the effective acoustic length is a (weak) function of frequency. Consequently, each of the acoustic tract resonances gives a slightly different effective length, with a range from 155 to 195 mm with glottis closed. Compared with the 1:3:5:7 ratios expected for cylindrical geometry, the higher resonances have slightly lower frequencies. This is perhaps because the cross-section in the region of the tract closer to the lips is on average greater than that of the region from the palate to glottis. However there is agreement between the length derived from the first acoustic resonance and the smoothed airway centroid length in the MRI of the mid-sagittal plane.

Speech Communication:
Paper ICA2016-586

Influence of changes of the glottal waveform on vowel production

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Abstract
Conditions of the vocal folds and upper airways can directly influence the fundamental frequency of the periodic movement of the glottis as well as the waveform of the source signal. This could further impair a patient's ability to excite resonances of the vocal tract and generate vowels. In this study, the Rosenberg model for the glottal pulse is applied to numerically investigate the propagation of the voice source signal from the glottis through a static vocal tract model. The geometries of the vocal tract are based on magnetic resonance imaging data for the different vowel pronunciations of a healthy male subject. For the computation of the pressure fluctuations and the associated distribution of frequency peaks as a result of the modulation through the vocal tract, direct compressible flow simulations are carried out by using a finite volume solver. The results are compared with the solution of a wave reflection analogue based on the area functions extracted from the same geometries and good agreement is reached. The effect of variations of glottal closure and fundamental frequency of the

standard Rosenberg waveform on the computed acoustic signal is investigated. Thus, an estimation of the impact of glottal diseases on the ability of vowel production is attempted.

Speech Communication:

[Paper ICA2016-699]

Copy synthesis of running speech based on vocal tract imaging and audio recording

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Abstract

This study presents a simulation framework to synthesize running speech from information obtained from simultaneous vocat tract imaging and audio recording. The aim is to numerically simulate the acoustic and mechanical phenomena that occur during speech production given the actual articulatory gestures of the speaker, so that the simulated speech reproduces the original acoustic features (formant trajectories, prosody, segmentic phonation, etc). The result is intended to be a copy of the original speech signal, hence the name copy synthesis. The shape of the vocal tract is extracted from 2D midsagittal views of the vocal tract acquired at a sufficient framerate to get a few images per produced phone. The area functions of the vocal tract are then anatomically realistic, and also account for side cavities. The acoustic simulation framework uses an extended version of the single-matrix formulation that enables a self-oscillating model of the vocal folds with a glottal chink to be connected to the time-varying waveguide network that models the vocal tract. Copy synthesis of a few French sentences shows the accuracy of the simulation framework to reproduce acoustic cues of natural phrase-level utterances containing most of French natural classes while considering the real geometric shape of the speaker. This is intended to be used as a tool to relate the acoustic features of speech to their articulatory or phonatory origins.

Speech Communication:

[Paper ICA2016-714]

Emotion recognition from speech using a physical model

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Abstract

In this paper, we present an emotion classification method for the estimation of human emotional states using speech. A 2-D emotional state is determined by two types of features. We recorded the utterances of an elderly person while they talked with an interviewer, who was a young university student. We manually labelled the subject’s speech based on the 2-D emotional model proposed by Russell et al. Utterances were manually classified as Active/Negative, Active/Positive, Passive/Negative, Passive/Positive, and Neutral. Emotional state transitions of the interviewee just before topic changes were analyzed. We then performed emotion classification experiments using a support vector machine (SVM). In addition to conventionally used acoustic features, we also added novel features derived from a two-mass, four tube vocal fold model, which included sequences of a subglottal pressure parameter, two kinds of stiffness parameters and three cross-sectional areas of the vocal tract, which were estimated from a vowel in each utterance using frames of 25 ms in length with a 10ms shift. We calculated the average, maximum, and minimum values of these features during utterances of a vowel, resulting in a total of 18 features. We tested the parameters using 166 emotional utterances (not including Neutral ones). As a baseline, 1,582 dimensional conventional acoustic features based on the spectral and prosodic characteristics of each utterance were extracted using the openSMILE toolkit. Features extracted using openSMILE achieved an f-measure rate of 0.549 during SVM classification experiments. When using only physical model parameters, a 0.354 f-measure was obtained, which is much higher than a chance rate. A combination of openSMILE and physical features achieved an f-measure rate of 0.564. Although the amount of improvement was not large, possibly because we only used one vowel from each utterance, we were still able to confirm that physical model parameters are effective for improving emotion recognition.

Speech Communication:

Paper ICA2016-662

Stimuli used in melodic intonation therapy with relation to autocorrelation function parameters

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Abstract

The presented work achieved a relationship between the stimuli used in Melodic Intonation Therapy (MIT), applied to speech recovery in patients who have suffered stroke, through listening, and statistical parameters obtained from the running autocorrelation function (r-ACF) parameters. To accomplish this, three stimuli for each level of the therapy were evaluated. All the stimuli were executed by twenty female speakers and twenty male speakers, all of them without speech problems. Then, these recordings were normalized, and the r-ACF parameters were calculated for each one. These parameters were statistically analyzed. Final results lead to conclusions for stimuli optimization.

Monday afternoon, 5 September 2016 **Room 204**

16:30 - 18:10

Speech Communication

SC1 - Speech Communication

Speech Communication:

Paper ICA2016-333

On the use of the Italian matrixed word tests in room acoustics for evaluating speech reception

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Abstract

This work aims to investigate the use of the newly developed Italian matrixed word test for room acoustics applications and assess the speech reception performance in complex acoustical environments. The detrimental effects of background noise and reverberation were taken into account and the listening experience was evaluated with both intelligibility scores and response times; the combined metric “listening efficiency” was also calculated, as the ratio of accuracy and latency measures. Twenty-one native Italian speakers were involved in the experiment, all of them reporting normal hearing. The listening tests were proposed in the closed-set format. SNR and reverberation were combined to create four different listening conditions, presented in a silent room via a three-dimensional audio rendering system. A stationary speech shaped noise and a non-stationary but continuous (ICRA) noise were used as maskers, aiming to cover different aspects of speech reception in noise (energetic masking and listening into the gaps). The results show that when reverberation is added to non-stationary noise, the fluctuating masking benefit is substantially reduced; the differences in accuracy and cognitive load that are present between the noises in anechoic conditions, are absent in a reverberant condition with T=0.94 s. A too long reverberation acts differently upon speech reception depending on the background noise, whereas a noise level increase of 3 dB similarly impairs the accuracy performance for both background noises.

Speech Communication:

[Paper ICA2016-357]

Effect of bandwidth on the intelligibility of Spanish words filtered through a series of pass-band filters

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Abstract
This paper investigates the speech intelligibility of the Argentinian-Castilian language in relation with its spectral information. A group of words was filtered through a series of pass-band filters with a slope of 4800 dB/oct. In order to clarify the relationship between the bandwidth of the filters and the intelligibility of the words, a subjective test was conducted with normal hearing people. To compare the results with the previous studies on English language, four different bandwidths (10, 20, 40, and 60 Hz) of the filter were used. As expected, a wider bandwidth of the filters improved the comprehension of the words filtered. A maximum percentage of the intelligibility reached 66.2% at a bandwidth of 60 Hz, much lower than that of the previous studies on English language.

Speech Communication:

[Paper ICA2016-721]

Large-scale analysis of Spanish /s/-lenition using audiobooks

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Abstract
Given forced alignment and accurate automatic phonetic classification and measurement, audiobooks are an important potential source of large-scale evidence about phonetic variation. For example, the audiobook version of the novel La Casa de los Espiritus, read by two Chilean actors, presents 17 hours of audio containing nearly 68,000 /s/ segments, distributed in a natural way across a wide variety of prosodic, lexical, morphological, syllabic, and phonetic environments. Thus we believe that this one audiobook offers more /s/ tokens than have been examined in the entire 50-year history of sociolinguistic study of Spanish /s/-lenition – and analysis on this scale allows statistical evaluation of a much larger set of hypotheses about phonetic variation and its conditioning factors. For broad comparison of geographical variants, we can use audiobooks whose readers exhibit a variety of accent types, in this case comparing works read by Chilean, Argentinian, Caribbean, Mexican, and Peninsular speakers. Most of the sociolinguistic literature on variation in Spanish syllable-final /s/ treats it as involving three distinct categories: retained [s], aspirated [h], and deletion. In our data we see coherent gradient variation in the duration and frication strength of /s/, with aspiration and deletion as continuum endpoints. Large-scale data also allows us to argue for cases of allomorphy, i.e. variable lexicalization of particular forms. In addition, we see several types of outcome not usually described, including the variable interpenetration of frication or aspiration with voiced portions of adjacent vowels or with following consonants, often resulting (for example) in breathy-voiced nasals.

Speech Communication:

[Paper ICA2016-847]

Speech recognition through the analysis of spoken syllables using autocorrelation function parameters

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Abstract
The presented work aims to establish a relationship between the identification of spoken syllables, using statistical parameters obtained from the running autocorrelation function (r-ACF) parameters. To accomplish this, six different syllables were recorded by twenty female voices and twenty male voices.

These recordings were normalized, and the r-ACF parameters were calculated for each file in order to statistically analyse them. Final results show a significant relationship between r-ACF parameters and the frequency spectrum of the vowels, which can lead to consonants classification according to pitch and loudness characteristics, regardless the speaker.

Speech Communication:

[Paper ICA2016-722]

Robust tonal and noise separation in presence of colored noise, and application to voiced fricatives

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Abstract

This study presents a method for separating periodic and aperiodic components embedded in speech signals. The fundamental frequency is first estimated from a frequency based technique using a whitened cumulative periodogram. A simple partial detector is used to avoid octave errors. The pitch detection is robust with respect to high level of colored noise. The periodic component is then estimated via the projection of the signal on the subspace spanned by the harmonics. The aperiodic component is obtained by subtracting the periodic component to the analyzed signal. Numerical validations on synthetic signals show that the presented method successfully separate the periodic and aperiodic components of simulated voice segments, even in very complicated case, such as voiced fricatives, which exhibit low and frequency-dependent harmonics-to-noise ratio. Applications on real speech signals highlight the interest of the technique to quantitatively estimate speech features such as harmonics-to-noise ratio, or voicing degree, as a function of time.

Monday afternoon, 5 September 2016 **Microcinema**
14:30 - 16:10
Animal Bioacoustics
AB1 - Animal Bioacoustics

INVITED

Animal Bioacoustics:

[Paper ICA2016-96]

Song and genetic divergences between migratory and sedentary populations of a songbird, the blackcap *Sylvia atricapilla*

Thierry Aubin[(a)], **Juliette Linossier**[(a)], **Sandor Zsebök**[(b)], **Emmanuelle Baudry**[(c)], **Hélène Courvoisier**[(a)]

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Abstract

In songbirds, songs are learned and involved in sexual selection. The cultural transmission of songs leads to dialects between populations and ultimately to speciation. Many songbirds migrate and individual differences in migratory patterns can influence population genetic structure and boost song differentiation. The complex interactions between song structures, migratory routes and genetic diversity remain to be understood. Blackcaps exhibit versatile songs with geographical variations and show a diversity of populations from sedentary to migratory. This species appears as a good model to study the relationships between migratory patterns, song variability and genetic diversity. Two populations were studied: a migratory population (2 groups around Paris) and a sedentary population (3 groups in Corsica). Studied individuals were ringed, blood samples were taken to study genetic relatedness and a detailed song analysis was performed. The complexity of the syllable repertoire

(>100) required the development of semi-supervised methods to classify syllables and of a custom-made program to compare sequences of syllables. Our analyses show that migratory birds have a greater syllable repertoire and a smaller repertoire of shared sequences compared to sedentary ones. However, although the turnover of individuals is higher among migrants than among sedentary birds, the 2 populations have similar syllable and sequence sharing within groups. Genetic analyzes with microsatellites loci show no genetic structure of groups and populations: individuals belonging to a same population are not genetically closer than those from different populations. Thus, it appears that in blackcaps, song dialects do not act as barriers to gene flows and are strongly cultural rather than genetic.

Animal Bioacoustics:

Paper ICA2016-908

Phonotactic response depends on trackball surface texture in *Gryllus bimaculatus* (Gryllidae, Orthoptera)

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Abstract

Mating behaviour in crickets is driven by acoustic communication. Phonotaxis is the behavioural process in which females are attracted and orient to male calling songs. We tested the female phonotactic response when the animals were walking on trackballs with different surface textures. Textures were measured with profilometry and were characterised as smooth, medium, or rough, with pore sizes of ~150, ~460, and ~800 micrometer, respectively. Female crickets walk better and have a higher phonotactic response on a rough or medium trackball surface, with numerous and large pores. A smooth surface, with small or few pores, prevents female crickets from walking properly, resulting in a significant decrease of their phonotactic response. Cricket claws are crucial for walking. Crickets hold on to the trackball by inserting their claws into the surface pores. If the surface is smooth or slippery, the crickets slide their feet and claws over the surface but cannot make proper mechanical contact. These findings may inform other studies that use trackball or treadmill systems, or arena experiments. The surface on which crickets are walking is crucial to obtain the optimal phonotactic response in behavioural studies.

Animal Bioacoustics:

Paper ICA2016-772

Passive acoustic monitoring of cetaceans as a tool for monitor the presence of cetaceans during seismic surveys

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Abstract

Potentially detrimental effects of marine seismic exploration upon marine mammals range from displacement from feeding or breeding areas, to auditory damage and potential mortality. Several nations where there are high levels of geophysical activity recognised the potential for such impacts and have formulated guidelines that aim to minimize potential detrimental effects of seismic surveys upon cetaceans. Brazilian guidelines are amongst the most restrictive and currently are the only ones that stop production if a dolphin or other small toothed whale is acoustically detected inside the exclusion zone. Analyses of the first years of experiences in Brazil showed that the acoustic detection of dolphins had the highest rate of shutdown, comparing to whale detections and visual observations. However, the estimated distance to small toothed whales is a challenge during real time mitigation since most parts of their vocal repertoire are composed by very directional (e.g., 5°), high frequency (e.g., 170 kHz), low duration pulses (e.g., 100 µs). The aim of this article is to discuss the Brazilian guidelines comparing it to others worldwide, with emphasis in the passive acoustic monitoring

methods used to estimate the distance to animals. In this article the detection methods and the necessity of its improvement will be discussed, attention will also be drawn to the necessity of an official examination by regulatory agencies**.**

Animal Bioacoustics:

Paper ICA2016-866

Development of a computational tool to analyze sounds: A biological study with anuran

Carolina Salgado Costa[a], Francisco Cantillo[b], Federico Iasi[b], Nilda Vechiatti[b], Guillermo Natale[a]

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Abstract

The emission of underwater sounds in anuran tadpoles has been documented in only two species from Argentina and one from Madagascar. Underwater sound emission by *Ceratophrys ornata* (Anura: Ceratophryidae) tadpoles was a novel finding reporting the first evidence in anuran larvae. The sound has been described as part of an antipredator mechanism that diminishes the frequency of predation between conspecifics. The aim of the study was to describe sound variability from tadpoles to adults with a novel technique in bioacoustics. Sounds emitted by tadpoles were both recorded underwater and out of water. The recording system consisted of a microphone and an interface coupled with a laptop. It was first calibrated with an acoustic reference source and compensated in order to obtain a recording system with flat frequency response. Audio recordings were digitalized, and post processed by a computational tool specifically developed in a numerical environment software. Variables selected to describe basic structure of sounds were: sound duration, number of pulses, number of inter pulses and dominant frequency. These variables were supplemented with typical acoustic parameters, such as equivalent continuous pressure level, peak sound pressure level, and spectral analysis with constant bandwidth filters. The interdisciplinary experience allowed developing a reliable system of recording, analysing thousand of sounds in a short period and therefore characterizing the sound of a species considering all variability.

Animal Bioacoustics:

Paper ICA2016-820

How echolocating bats listen to their echoes

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Abstract

The echolocating bats emit ultrasonic pulses and listen to echoes to catch preys and measure characteristics about their environment during their flight. It has been known that they can precisely measure these in real time. However, returning echoes from small objects are scattered and attenuated easily. We have conducted experiments with flying bats and non-flying bats to investigate how they extract information they need. They precisely detect preys and measure characteristics surrounding their environment. Findings have shown that the bats do not directly listen to the echoes reflected from a small insect but listen to echoes reflecting from a large stable object located far way, which contain information about a flying insect. Summarized data are discussed.

INVITED

Product Sound Quality and Multimodal Interaction:

Paper ICA2016-672

Structural analysis of the value evaluation of vehicle door-closing sounds

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Abstract

Vehicle door-closing sounds affect the commercial value of vehicles. To create a door-closing sound that adds vehicle value, it is necessary to grasp and control factors affecting the value evaluation, such as the sound quality and imagery associated with the door-closing sound. The present study conducts psychoacoustic experiments to comprehensively investigate factors affecting the value evaluation of door-closing sounds and its structure using the evaluation grid method. Pairs of door-closing sounds were presented to participants. They were asked to select the more satisfactory stimulus and to give reasons for their selection so as to clarify the "original evaluation factors" (i.e., perspectives in evaluation). Furthermore, to elicit factors relevant to the obtained evaluation factors at the higher/lower levels of the hierarchical structure, participants were asked to identify benefits (i.e., reasons why they provided the original evaluation factors) and their detailed concrete requirements (mainly acoustic characteristics of stimuli). The overall structure was obtained from relationships between original evaluation factors and elicited factors at the higher/lower levels for all paired comparisons of door-closing sounds of all participants. The results reveal that door-closing sounds with abundant low-frequency contents and rapidly damped energy aroused feelings of massiveness and door closing. Furthermore, these feelings were related to emotional benefits such as a sense of security and an impression of the luxuriousness of the vehicle. Relationships between ratings of paired comparisons and metrics confirmed the effects of acoustic features of door-closing sounds found in the structure. The results suggest factors important to the design of door-closing sounds.

INVITED

Product Sound Quality and Multimodal Interaction:

Paper ICA2016-557

A sound quality model for washing machine sounds based on artificial neural network

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Abstract

The sounds of the washing machines are one of the most complicated household product sounds. They are instationary and contains different operating stages which cause totally different sounds. These aspects make the psychophysical evaluation and the modelling of the overall washing machine sound quality very difficult. Washing and spin are two main operational stages. Particularly spin sounds are loud and highly instationary with various sound events. The aim of this study to predict the sound quality perception of front-loading washing machines using artificial neural networks. Therefore a listening test was conducted, in which the participants evaluated the pleasantness/annoyance of the washing machine sounds. Then the psychoacoustical and signal features of the sounds were analysed. The link between the listening test results and analysis results were realized using artificial neural networks (ANNs). Finally the model which is based on the ANNs was verified using test stimuli. The results of the study was compared with the models from our previous studies.

Product Sound Quality and Multimodal Interaction:
Paper ICA2016-113

Modelling the sensation of fluctuation strength

Alejandro Osses Vecchi[a], **Rodrigo García León**[a], **Armin Kohlrausch**[a;b]
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Abstract
The sensation of fluctuation strength (FS) is elicited by slow modulations of a sound, either in amplitude or frequency (typically < 20 Hz), and is related to the perception of rhythm. In speech, such periodicities convey valuable information for intelligibility (prosody). In western music, most of the envelope periodicities are also found in that range. These are evidences of the potential relevance of FS in the perception of speech and music. There is, however, no published computational model to assess the FS of a sound. This might be one reason why when slow modulations of a sound are to be analysed, other indirect measures (e.g., loudness to estimate "loudness fluctuations") or more complex techniques (e.g., the modulation filter bank) are used. In this paper we present a model of fluctuation strength. Our model was developed taking advantage of the physical similarity between FS and the psychoacoustical sensation of roughness. The FS model was then adjusted and fitted to existing experimental data collected using artificial stimuli, namely, amplitude- (AM) and frequency- (FM) modulated tones and amplitude-modulated broadband noise (AM BBN). The test battery of sounds also considered samples of male and female speech and some musical instrument sounds.

INVITED

Product Sound Quality and Multimodal Interaction:
Paper ICA2016-558

The quality of potato chip sounds and crispness impression

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Abstract
The acoustical signals are information carriers and the sound of a product is a hint of its quality. We communicate with industrial products in various situations and their sound inform us about the product, its operating condition or its quality. In some cases the product sounds can evoke emotional associations. For example, the roaring sound of a vehicle can be associated with the sportiness or the rattle of an oldtimer can be associated with the nostalgia. Not only the sounds of the industrial products, but also the sounds of the food deliver us various information. Earlier life (eating) experiences play an important role on this issue. We learn in our early childhood the connection between the acoustical parameters and the nutritional properties. The objective of this study is to investigate the relationship between the properties of the chip bite sound and the perceived crispness. In an experiment, the crispness of chip sounds were evaluated. In this experiment, recordings of the sound of 5 chips and filtered variations of the recordings were presented to the subjects. Then, a link between the perceptually important signal properties and the crispness was established.

Product Sound Quality and Multimodal Interaction:
Paper ICA2016-907

Research on nonlinear evaluation model of cooling fan sound quality

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Abstract
The cooling fans are usually used for cooling machines to keep them running well. However, they also cause noise and do harm to users' hearing system, nervous system, even cardiac and cerebral functions. Aiming at providing evaluation criterion to noise deduction by predicting the human's feeling about noise, this paper established a nonlinear evaluation model of cooling fans' sound quality. 13

sound samples of cooling fans were collected with HEAD Recorder and HMS IV. After editing, 33 samples were saved with the duration of 5s. In the subjective evaluation experiment, 30 subjects were recruited to mark each sound sample. For objective evaluation, the 33 samples were imported into ArtemiS to analysis psychoacoustic parameters, Sound Pressure Level (SPL) and A-weighed Sound Level. A Correlation Analysis was done first to screen parameters roughly and tonality was removed with low correlation coefficient of -0.493. Then research did Linear-regression Analysis and screened parameters strictly. According to the Run Test results of Standardized and Studentized Residual, the Asymptotic Significances (2-tailed) were both less than 0.05 which indicates that residuals were not mutually exclusive and this model was not functional. The model was repeatedly modified until the Asymptotic Significances (2-tailed) became greater than 0.05. A nonlinear evaluation model was established finally with independent variables of loudness, sharpness, SPL and A-weighed sound level.

Monday afternoon, 5 September 2016 **Auditorium 2**
14:30 - 16:10
Numerical Techniques
NT1 - Boundary Element and Meshless Methods on Acoustics and Vibrations

INVITED

Boundary Element and Meshless Methods on Acoustics and Vibrations:
Paper ICA2016-864

RBF-based shapes optimized with genetic algorithms for sound diffusion

Ricardo Patraquim[a], Luís Godinho[a], Paulo Amado-Mendes[a]
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Abstract
Development of sound diffusion technical solutions has been a topic of intense research in the last years. Many diffuser shapes, based on mathematical series or in different optimization techniques, have been suggested. In this paper, the authors propose an alternative technique to define new shapes of sound diffusion configurations, based on the use of radial basis functions (RBF). In addition, to allow the definition of optimal surface shapes for a given frequency band, a genetic algorithm is used. The diffusion coefficient is computed within the optimization procedure using the Kirchoff integral equation. The global procedure presented here allows simple organic shapes to be obtained, which are of significant interest in architectural design of acoustic spaces.

INVITED

Boundary Element and Meshless Methods on Acoustics and Vibrations:
Paper ICA2016-19

The MFS as a tool for the numerical analysis of vibration protection devices

Carlos Albino[a], Luís Godinho[a], Daniel Dias-da-Costa[b], Paulo Amado-Mendes[a]
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Abstract
Buried structures may be used to control elastic wave propagation in soils and help reducing vibrations in sensible structures. The analysis of these effects using numerical tools is of high importance and is usually a demanding computational task. In the present work, the authors analyse the possibility of using a meshless method for such simulations, which is known as the Method of Fundamental Solutions (MFS). In many applications, the MFS has proved to be a worthy and more efficient alternative to classic methods, such as the BEM or FEM. Here, the authors present a detailed numerical study on the performance of the MFS to simulate the propagation of elastic waves in a soil

with multiple buried inclusions. An application example is also presented, in which a decoupled numerical procedure is used to analyse the vibrations induced by a dynamic load in a building structure when inclusions buried in the soil act as a vibration shielding barrier.

Boundary Element and Meshless Methods on Acoustics and Vibrations:
Paper ICA2016-181

Boundary element method for acoustic radiation force and torque acting on non-spherical particles

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Abstract
Ultrasound has been used to manipulate micro particles, such as biological cells, in microfluidic devices. To design microfluidic devices capable of sorting and trapping specific micro particles, the acoustic radiation force and torque exerted on these micro particles need to be calculated accurately. We have developed a boundary element formulation to calculate the force and torque acting on particles of arbitrary shape, size and orientation with respect to the ultrasound field in a microfluidic channel. This provides a more versatile and accurate calculation of force and torque over analytical solutions that are available only for simple shapes, such as spheres and ellipsoids, typically in an axisymmetric configuration. The first order acoustic scattered field from the particle is first solved using the boundary element method. The incident and scattered fields are then approximated using regular and multipole expansion coefficients, respectively, at the centroid of the particle. Lastly, the radiation force and torque acting on the particle are calculated based on the interaction between the coefficients of the scattered and incident fields. Using this formulation, the force and torque acting on non-spherical particles are calculated. Parametric studies on the effects of particle size, shape, and orientation to the incident field will be reported.

Boundary Element and Meshless Methods on Acoustics and Vibrations:
Paper ICA2016-309

Efficient calculation for evaluating vast amounts of quadrupole sources in BEM using fast multipole method

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Abstract
There are increasing demands for computational prediction of the propagation of flow-induced noise. As numerical approaches for predicting flow-induced noise, finite-difference method (FDM), finite-element method (FEM) and boundary-element method (BEM) are extensively used to solve the Lighthill's equation or the Curle's equation. Among these approaches, the BEM has a wide field application due to several benefits such as smart modeling of the acoustic radiation field and easy mesh generation. Despite these benefits, both memory requirement and calculation complexity increase by the second power of the number of DOFs in the BEM approach. Therefore the BEM with the application of the fast multipole method (FMBEM) was developed. The FMBEM reduces both memory requirement and calculation complexity to the linear increase. However, when BEM is applied to predict the propagation of flow-induced noise, calculation cost for evaluating quadrupole point sources becomes to be unpractical level. This is due to the fact that the effect of each source should be evaluated at each boundary element in the BEM procedure. Therefore, the calculation complexity and memory increase by the factor of the number of quadrupole sources times the number of boundary elements. To reduce the calculation complexity and memory, the fast multipole method is applied for the quadrupole sources evaluation. Consequently evaluation time was reduced to almost

linear manner. In this paper, the analysis method, the validation of various parameter settings and some numerical examples are shown.

Boundary Element and Meshless Methods on Acoustics and Vibrations:
Paper ICA2016-418

Efficient boundary element analysis of periodic sound Scatterers

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Abstract
A boundary element technique is used to formulate exterior acoustic problems comprising of periodic arrangements of sound scatterers. The matrix equation formulated by the boundary element method for this acoustic scattering problem is a block Toeplitz matrix. The discrete Fourier transform is then employed in an iterative algorithm to solve the block Toeplitz system. Solving a periodic acoustic problem using the block Toeplitz system significantly reduces computational time and storage requirements. Solid cylindrical scatterers in a periodic square lattice arrangement are examined. Result for the insertion loss of the sonic crystal barrier is presented. Directivity and contour plots of the total acoustic field at selected discrete frequencies are also presented and compared with those obtained by the finite element method.

Monday afternoon, 5 September 2016 **Auditorium 2**
16:30 - 18:30
Numerical Techniques
NT2 - Numerical Techniques (others)

INVITED
Numerical Techniques (others):
Paper ICA2016-802

Assessment of the far-field sound radiation of ducts using the Lattice Boltzmann method and a two-dimensional FFOWCS Williams and Hawkings formulation

Danilo Braga[(a)], José Santana Neto[(a)], André Spillere[(a)], Andrey Ricardo Da Silva[(a)], Júlio Cordioli[(a)]
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Abstract
The characteristics of the acoustic far-field radiated by a duct highly depends on the geometric aspects found at its open end. This work proposes a simple numerical technique in order to investigate the parameters associated with normal mode radiation of ducts issuing a subsonic mean flow. The technique is based on a hybrid approach involving the lattice Boltzmann method and the Ffowcs Williams and Hawkings formulations for a porous surface. The results are presented in terms of reflection coefficient and sound directivity and compared with the exact analytical solutions provided in the literature for low subsonic mean flows. The good agreement between numerical and analytical solution suggests that this approach can be used to investigate the duct's open end geometry on the parameters associated with sound radiation.

Numerical Techniques (others):

Paper ICA2016-101

Auralization of road traffic/construction noise by using numerical analysis method with digitized convolution technique

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Abstract

The auralization of noise prediction results, such as for road traffic or construction, is useful in order to grasp the actual status of noise or evaluate various noise countermeasures. In this paper, we report on an auralization method based on analysis of acoustic waves, which used the discrete impulse response as the initial condition for calculation. we propose boundary conditions for the incident wave to the analysis area, which gives appropriate physical values for sound propagation from a point sound source to the grid points on the boundary. It is possible to reduce the computation costs, as it is not necessary to prepare grid points around the sound source. Furthermore, by using this virtual reality technique, we present some benchmarks for investigating the validity of the auralization for construction noise. The sound convolved by dry sound with the numerical results of the discrete impulse based on the convolution quadrature is emitted in the VR space.

Numerical Techniques (others):

Paper ICA2016-561

Optimised 25-point finite difference schemes for the three-dimensional wave equation

Brian Hamilton[(a)], **Stefan Bilbao**[(a)]
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Abstract

Wave-based methods are increasingly viewed as necessary alternatives to geometric methods for room acoustics simulations, as they naturally capture wave phenomena like diffraction and interference. For methods that simulate the three-dimensional wave equation—and thus solve for the entire acoustic field in an enclosed space—computational costs can be high, so efficient algorithms are critical. In terms of computational complexity, finite difference schemes are possibly the simplest such algorithms, but they are known to suffer from numerical dispersion. High-order and optimised schemes can offer improved numerical dispersion, and thus, computationally efficient numerical solutions. In this paper, we consider two families of explicit finite difference schemes for the second-order wave equation in three spatial dimensions, using 25-point stencils on the Cartesian grid. We review known special cases that lead to high-order accuracy in space (and possibly in time), and we present new schemes with optimised stencil coefficients. These schemes provide accurate wave simulation using substantially less memory than the conventional scheme. Simulations are presented to demonstrate the performance of the optimised schemes.

Numerical Techniques (others):

Paper ICA2016-736

Passive time-domain numerical designs for room acoustics simulation

Stefan Bilbao[(a)] , **Brian Hamilton**[(b)]
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Abstract

The design of stable time domain numerical simulation methods for room acoustics simulation is a challenging problem. One chief difficulty is in the determination of appropriate stable boundary

terminations, particularly when the room geometry is irregular, and when the wall condition is spatially-varying and/or frequency-dependent in a non-trivial way. In this paper, design strategies for stable simulation are presented, based on the finite volume time domain method (FVTD), which, due to its unstructured character, allows for flexible modelling of irregular room geometries. Furthermore, FVTD reduces to the popular finite difference time domain (FDTD) method under certain choices of regular structured mesh. Under locally-reactive wall conditions, the boundary condition can be characterised by a positive real admittance function, variable over the extent of the room boundary. Using frequency-domain analysis techniques, it can be shown that solutions to the complete system are non-increasing. Furthermore, such analysis techniques can be extended to the case of discrete time simulations, leading to numerical stability conditions for a complete room simulation. Distinct explicit and implicit time-domain simulation methods are analysed in this manner. Extensions to the case of non-locally reactive conditions are discussed.

Numerical Techniques (others):

Paper ICA2016-210

A boundary integral operator method for modelling uncertainties in vibro-acoustics

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Abstract

Dynamical Energy Analysis (DEA) is an approach for studying the vibro-acoustic response of complex systems in the high frequency limit. The method has been extended to industrial scale applications using an efficient implementation on meshes known as Discrete Flow Mapping. DEA is a deterministic boundary transfer operator method for the modelling of phase-space densities (or ray densities) arising in the ray-tracing approximation of a linear wave problem. In this work, we investigate extensions of the DEA approach to stochastic boundary transfer operator methods by replacing the deterministic description of the ray flow with a probabilistic flow map incorporating various sources of uncertainty. We will present efficient numerical approaches with relevance to high-frequency vibro-acoustic wave problems.

Numerical Techniques (others):

Paper ICA2016-426

Advances in the holistic numerical simulation workflow to analyze the sound of combustion engines based on human auditory perception

Fabian Duvigneau[a], Ulrich Gabbert[a]

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Abstract

In this paper a holistic simulation workflow is presented, which aims to calculate the resulting sound radiation of combustion engines with only the cylinder gas pressure curve as input data. The crank drive motion is analyzed with the help of an elastic multi-body model which also takes into account the elasto-hydrodynamic interactions in the fluid films. The multi-body model is coupled with a finite element model of the crankcase and its mounted parts, which allows the surface velocity to be calculated. This model is also coupled with a finite element based acoustic model of the ambient air volume in which the pressure distribution at any point in the acoustic fluid can be calculated. Finally, in the last step of the workflow the acoustic results are evaluated with respect to human auditory perception through a complex psychoacoustic model. Therefore, a listening test has to be carried out in advance in order to generate the psychoacoustic model. It should also be noted that the psychoacoustic model could be reused in future applications if it is a sufficiently similar configuration with respect to that used for generating the psychoacoustic model. Recently, the overall simulation workflow has been improved by taking into account the influence of the motor oil on the resulting perception of the auralized simulation results in a computationally efficient way. Furthermore, a technique inspired by the MP4-technology is implemented in the auralization component of the existing holistic workflow to further increase the overall efficiency of the process.

Phononic Crystals and Acoustic Metamaterials:
Paper ICA2016-74

Dual-band negative index ultrasonic metafluids

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Abstract
The extraordinary properties of acoustic (random) metamaterials, such as negative refractive index, originate from low frequency resonances of sub-wavelength particles. While most of these functional materials are fabricated by mechanical engineering, soft matter techniques coupled with microfluidics open a new synthesis route for acoustic metamaterials [Brunet *et al*., *Science* 342, 323-324 (2013)]. As a demonstration, we have achieved soft 3D ultrasonic metafluids with negative index composed of large amounts of calibrated soft porous micro-spheres, acting like strong Mie resonators [Brunet *et al*., *Nature Materials* 14, 384-388 (2015)]. The wide variety of physico-chemical processes offered by chemical engineering allows for the full-control of the mechanical/acoustical parameters (elasticities/celerities) of these resonant micro-particles. To illustrate the strength of our "soft" approach, we have recently shown that it is not only possible to achieve soft 3D ultrasonic metafluids with one negative band [Raffy *et al*., *Advanced Materials* 28, 1760-1764 (2016)], but also with two separate ones [3]. The emergence of the second negative band is due to shear waves that propagate within the resonant micro-beads. If often neglected in most theoretical works, shear wave may induce a dipolar (transverse) resonance that leads to a negative index when it overlaps the monopolar (longitudinal) resonance. Finally, the Poisson coefficient, which parametrizes the ratio of transverse-to-longitudinal sound celerities, will be shown to be a relevant mechanical parameter to anticipate whether or not the second negative band will emerge.

Phononic Crystals and Acoustic Metamaterials:
Paper ICA2016-579

Multiple scattering enables negative index in single negative metamaterials: proof with an acoustic superlens

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Abstract
Metamaterials are composite media structured at a scale much smaller than the wavelength and offer incredible possibilities to engineer the propagation of waves. Here, we verify through numerical simulations that negative index acoustic metamaterials based on Helmholtz resonators only is realizable, thanks to the muliplescattering occuring at the subwavelength scale. We propose an experimental demonstration of a negative index acoustic superlens, achieving subwavelength focusing and imaging with spot widths and resolutions respectively 7 and 3.5 times better than the diffraction limit. This has profound implications concerning the physics of metamaterials and introduces the concept of metamaterial crystals.

Phononic Crystals and Acoustic Metamaterials:

Paper ICA2016-104

Extremely slow edge waves in mechanical graphene with rotating grains

Li-Yang Zheng[a], **Vincent Tournat**[b], **Georgios Theocharis**[c],**Vitalyi Gusev**[d]

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Abstract

Edge elastic waves are investigated on both zigzag and armchair boundaries of a mechanical granular graphene where spherical beads are assembled in a single-layer honeycomb structure. Due to the existence of the rotational degrees of freedom of the grains, rotation-associated shear, bending and torsional couplings between the neighbor beads are activated. The dispersion curves of edge waves are theoretically derived and numerically evaluated. In particular, the influence on the edge waves of weak interactions between the beads through the bending and torsional moments is studied. Quasi-flat branches with near zero frequency are revealed. These bands, supporting the propagation of edge waves with extremely slow group velocity, transform into the perfect zero-frequency bands for zero torsional rigidity or vanish for zero bending rigidity, indicating that weak bending and torsional inter-grain interactions are playing a crucial role in the existence of extremely slow modes. The velocities of the extremely slow propagating modes are controlled by the effective bending and torsional rigidities that are much softer than normal and shear rigidities of the intergrain contacts ensuring the propagation of much faster waves, for example those existing when the rotational degrees of freedom are blocked, which are themselves much slower than the acoustic waves that could propagate in the material composing individual grains. The investigation on edge waves in mechanical granular graphene with rotational degrees of freedom is the necessary preliminary step for the design of the granular meta-graphene with an artificially broken/modified symmetry for inducing topologically protected unidirectional edge states or other functionalities.

Phononic Crystals and Acoustic Metamaterials:

Paper ICA2016-368

Symmetry breaking and topology of acoustic waves in phononic structures

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Abstract

We present examples of phononic structures that beak four types of symmetry, namely timereversal symmetry, parity symmetry, chiral symmetry and particle-hole symmetry. The implications of symmetry breaking on the topology of the acoustic wave function in the space of its Eigen values are discussed. Particular attention is focused on the torsional topology of acoustic waves in periodic media in wave vector space. Two types of approach to achieve symmetry breaking are considered: (a) intrinsic topological phononic structures whereby symmetry breaking occurs from the internal structural characteristics, and (b) extrinsic topological phononic structures where external stimuli such as spatio-temporal modulations of the physical properties of the medium are used to break symmetry.

Phononic Crystals and Acoustic Metamaterials:
Paper ICA2016-61

Non linear wave phenomena in piezoelectric phononic crystals with time-dependent electric boundary conditions

Charles Croënne[(a)], Olivier Bou Matar[(b)], Jérôme O. Vasseur[(c)], Marie -Fraise Ponge[(d)], Anne-Christine Hladky-Hennion[(e)], Pierre A. Deymier[(f)], Bertrand Dubus[(g)]
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Abstract
This work focuses on elastic wave propagation in phononic crystals constituted by homogeneous piezoelectric materials with periodic distributions of electrodes. In such periodic structures, dispersion curves are strongly modified by the variations of the external electrical impedances connected to the electrodes. The case of elastic pulses interacting with such piezoelectric phononic crystals with time dependent electrical boundary conditions is considered. Simulations are performed using a specific Finite Difference Time Domain model developed for the one-dimensional case. Two configurations of time-dependent periodic electrical boundary conditions on the electrodes are studied: i) switch from floating to grounded electrodes during a finite time window; ii) grounded electrodes moving at constant speed. For each case, non linear effects in pulse transmission through a phononic crystal slab are analysed.

Monday afternoon, 5 September 2016 — **Auditorium 3**
16:30 - 17:50
Physical Acoustics
PA1 - Phononic Crystals and Acoustic Metamaterials

Phononic Crystals and Acoustic Metamaterials:
Paper ICA2016-620

Geometric frustration and band gaps in acoustic networks

Pai Wang[(a)], Yue Zheng[(b)], Matheus C. Fernandes[(a)], Yushen Sun[(c)], Kai Xu[(d)], Sijie Sun[(a)], Sung Hoon Kang[(e)], Vincent Tournat[(a; f)], Katia Bertoldi[(a;h)]
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Abstract
We present a theoretical, numerical and experimental study of the effect of geometric frustration on the propagation of sound waves in 2D macroscopic acoustic networks comprising a periodic array of connected waveguides. Interestingly, we show that in networks of triangular and pentagonal geometry, the existence of band gaps can be interpreted in the frame of frustration, revealing a different mechanism than usual scattering to suppress the propagation of pressure waves in specific frequency ranges. Thus, analogously to other natural or artificial configurations, liquid crystals, proteins, water ice, the concept of geometric frustration can be extended to acoustic networks of tubes. These results provide a basis for the design of acoustic lattices of connected tubes with increasing levels of complexity, and possibly comprising local resonators or other acoustic elements, in order to be able to fully engineer and tune the dispersion in two or even three dimensions.

Phononic Crystals and Acoustic Metamaterials:

Paper ICA2016-345

Frequency-dependent dissipation in dispersive wool felt

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Abstract

Felt is a non-woven fabric (textile) that is produced by matting, condensing and pressing natural or synthetic fibres by a process called wet felting. Felt is the oldest form of fabric known to humankind by predating weaving and knitting. There are many different types of felts for industrial, technical, designer and craft applications. While some types of felt are very soft, some are tough enough to form construction materials. Felt can vary in terms of fibre content, dimensions, density and more factors depending on the use of the material. Not many physical properties of felt are well known or actively studied. The purpose of this study is to investigate compressional strain wave propagation through natural wool felt. In this paper a frequency-dependent dissipation and dispersion of acoustic waves propagating through felt are analysed in the onedimensional setting. The presented model is based on a experimentally obtained constitutive relation that takes into account the elastic and hereditary properties of the microstructured felt. The numerical solutions of the linear problem are used to estimate a strain pulse amplitude decay and they are analysed in the context of complex dispersion curves. It is shown that in the linear case the exponential decay rates for different frequencies may be obtained rather accurately by using dispersion analysis. It is concluded that intertwined and anisotropically oriented fibres in porous felt give rise to frequency-dependent attenuation of acoustic waves propagating through the material. Presented results are useful for various acoustical applications of felt material.

Phononic Crystals and Acoustic Metamaterials:

Paper ICA2016-402

Sound propagation in permeable materials with locally resonant elastic frame

Rodolfo Venegas[a]**, Claude Boutin**[b]
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Abstract

This paper investigates sound propagation in permeable materials with locally resonant elastic frame. An example of this type of materials is one having a permeable microstructure whose solid frame is made of a stiff skeleton on to which highly-flexible thin films are fixed. Conversely to the case of Biot poroelastic materials where the fluid flow is not affected by the frame deformation, the presence of the films can significantly alter the fluid flow through the material. As a consequence of the fluid-structure interaction, and in particular of the local resonances, the acoustic behaviour departs from the classical physics leading to the Biot description. To evidence this, the theory of homogenization for periodic media is used to derive the macroscopic description of sound propagation through this type of materials. The description is then asymptotically analysed to determine the conditions for which the local resonances affect the propagation of sound waves in the material strongly. Experimental validation of the theory and numerical simulations for different permeable locally resonant materials are also presented.

Phononic Crystals and Acoustic Metamaterials:

Paper ICA2016-896

Laboratory and full-scale experimental evaluation of the acoustic behaviour of sonic crystal noise barriers

Paulo Amado-Mendes[a], Luís Godinho[a], Pedro Gil Santos[a], Alfredo G. Dias[a], Mário Martins[b]

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Abstract

Sonic crystals represent a particular case of acoustic metamaterials, corresponding to periodically arranged structures, with individual scatterers regularly spaced. These structures present a particular acoustic behaviour, revealing high levels of sound attenuation in a range of frequencies known as the band-gap. By properly selecting some definition parameters of the sonic crystals, for example, among others, the type of lattice, the lattice spacing or the diameter of the scatterers, the observed band-gap can be adjusted to match a given frequency of interest. This idea was explored to study the use of timber logs, arranged in periodic ways, to build sustainable noise traffic barriers. In the present work, the different steps of the experimental evaluation, performed to validate this concept, are described and the results of some selected experimental cases are presented, corresponding to reduced-scale laboratorial setups and to a full-scale prototype analysed in outdoor conditions.

Tuesday, 6 September 2016

Tuesday morning, 6 September 2016 **Juan Pablo II Auditorium**
09:00 - 10:40
Environmental Acoustics & Community Noise
EN1 - Noise Assessment and Control

Noise Assessment and Control:
[Paper ICA2016-92]

Assessment of annoyance, noisiness and loudness caused by environmental noise sources

Nicolás Urquiza
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Abstract
In this research, the study of environmental noise in downtown Caseros, located in Buenos Aires Province, was addressed through the analysis of six sound events by objective and subjective methods. In order to do this, physical properties of each sound stimulus were evaluated and auditory surveys were conducted to study the effects of noise from the psychoacoustics point of view. Afterwards, the correlation between objective acoustical parameters and three subjective attributes (annoyance, noisiness and loudness) was investigated.To asses the subjective attributes three carefully designed surveys were conducted; an online pilot survey and two auditory surveys performed under laboratory conditions by paired comparison test and verbal rating scale method.Moreover, objective variables from each sound events were analysed through their frequency spectrums and acoustic descriptors. Afterwards, the correlation coefficients between objective parameters and the subjective response of people were studied. Results indicated that the most annoying sound events had tonal characteristics and loudness was the only subjective attribute that showed good correlation with several objective acoustic descriptors. In addition, surveys confirmed that pairwise comparisons and verbal rating scale method have excellent correlation for the same subjective attribute. Finally, it was concluded that is necessary to perform subjective studies in order to complement the purely objective measurements when the existence of annoyance in a population exposed to noise is evaluated.

Noise Assessment and Control:
[Paper ICA2016-67]

Reduction of noise annoyance through public participation

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Abstract
In a country such as Germany with a dense population and a high traffic volume, large parts of the population are affected by noise. In order to improve the noise situation significantly, all noise abatement measures have to be used. This includes the involvement of citizens in decision-making processes because it can diminish their annoyance reaction. A prominent example of public participation is a model project, which started in 2015 in the City of Leipzig. In this project, several local residents identify concrete traffic noise problems and make proposals for noise reduction. Measures that are feasible on a short-term basis are in the focus. For instance, a speed reduction from 50 to 30 kilometers per hour is discussed. Within the framework of the project, the management has commissioned an urban planning office, which advises the citizens on noise issues. Moreover, the process is accompanied by a committee, which consists of representatives of the administration of Leipzig, citizens associations, politicians and public transport operators. The structure of the project is pointing the way for future processes of public participation. The project shows that public participation leads to a better quality in results and improves people´s quality of life.

Noise Assessment and Control:

Paper ICA2016-666

Sound exposure measurements using hearing-aid technology

Simon Boelt Jensen[a], **Mads Drastrup**[b], **Esteban Chávez Morales**[c], **Rodrigo Ordoñez**[d], **Carsten Borg**[e]

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Abstract

Sound exposure is one of the primary causes of preventable hearing loss. Traditionally, sound exposure has been associated to industrial settings, and as such, treated as an occupational safety issue leading to international standards regulating sound exposure to improve working conditions. High levels of sound exposures are experienced in modern society in many different situations such as attending concerts, sport events and others. This leads to an interest in measurement devices which are discreet and simple to use, in order to assess sound exposures encountered in typical daily life scenarios. The purpose of this work is to document the use of a modified behind-the-ear (BTE) hearing-aid as a portable sound pressure level (SPL) meter. In order to obtain sound level measurements with a BTE device comparable to sound field values that can be used with existing risk assessment strategies, differences due to microphone positions and the presence of a person in the measurement must be taken into account. The present study presents measurements carried out to document the characteristics of the BTE device, using the same framework presented in the ISO 11904 standard series. The responses at the BTE position on a head and torso simulator (HATS) were measured and combined with the A-weighting filter, frequency weigted sound field values. The compensation filters improved the accuracy of the BTE devices especially in laboratory conditions. Field tests corroborate the necessity of both diffuse- and free-field compensation devices showing better approximations for corresponding sound field scenarios.

Noise Assessment and Control:

Paper ICA2016-56

Aircraft noise protection strategy in Germany

Roman Thierbach[a], **Renè Weinandy**[b], **Thomas Myck**[c]

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Abstract

Numerous people in Germany are affected by aircraft noise. Noise from aircraft operating in the surrounding of major airports is not only annoying for residents; it may also lead to serious health impacts. Therefore, the instruments and measures currently available to reduce aircraft noise still need to be used more effectively and in a more targeted manner. A significant reduction of noise emitted by air traffic can only be achieved by a comprehensive noise protection strategy. It comprises of a variety of elements from measures to limit the noise emission of aircraft and the use of modern flight operation procedures to economical instruments, land-use planning at airports and legal regulations, e.g. curfews during the night-time. All these measures are to be applied in a coordinated way in order to improve the noise situation at the airport. The various instruments and measures for aircraft noise abatement at airports in Germany will be discussed and evaluated.

Noise Assessment and Control:

Paper ICA2016-81

A five year follow-up: noise exposure and hearing loss in classical orchestra musicians

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Abstract

Noise exposure and hearing loss was assessed in different instrument groups of a professional ballet orchestra. Those group members experiencing the highest levels of exposure also had the highest pure tone thresholds. We found that thresholds were not uniform across instrument groups. The greatest difference in thresholds was observed at test frequencies above 2000 Hz, peaking at 4000 Hz where the average difference between groups was as high as 15 dB. Five years have elapsed since these initial measurements were taken. In this follow- up we reassess differences across the instrument groups in pure tone thresholds, and noise exposure. We also include a measure of functional hearing. This study provides information that extends current understanding of the occupational risks faced by professional musicians playing in orchestras.

Tuesday morning, 6 September 2016 — **Juan Pablo II Auditorium**
11:00 - 12:00
Environmental Acoustics & Community Noise
EN1 - Noise Assessment and Control

INVITED

Noise Assessment and Control:

Paper ICA2016-829

Noise generated from large construction sites: Measurements and possible mitigations

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Abstract

In this study, the issue of noise arising from the activities of large construction sites is analysed. The problem is particularly significant when the sites are close to protected natural areas or to residential areas and also when the duration of the construction works is very large. The case of the construction of a mobile barrier system, known as the MoSE, for the safeguard of the city of Venice from intruding tidal waters, is here reported. Since the early beginning of the construction activities, in April 2005, noise emission monitoring was conducted in order to evaluate possible effects on the presence of bird communities in the surrounding areas and also to evaluate the noise disturbance in some of the residential buildings close to the sites. During the monitoring activity, it was possible to measure the noise levels, the noise spectra and spectrograms in real-time and occasionally even the audio signal for particular activities was recorded. The analysis of data collected in different periods defined the pile driving and some other activities as the most significant from the standpoint of noise emission, therefore the possibility to mitigate their emission was investigated.

Noise Assessment and Control:

[Paper ICA2016-254]

10 lessons learned as an entry-level transportation noise control engineer

Priscilla Brownlow

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Abstract

Are you a student or recent graduate considering a noise control engineering career? A typical entry-level noise control engineer spends 17 years in school before entering the workforce; transitioning from academia to industry can be difficult. There are many subtle and not-so-subtle differences and pitfalls. Most importantly, your work gains real-world significance. These 10 lessons learned present some of my pitfalls encountered as an entry-level noise control engineer, what to expect in the office, field experiences, and professional challenges. Hopefully they will provide insightful advice to those of you preparing to step out of the classroom and into the 'real world'.

Noise Assessment and Control:

[Paper ICA2016-450]

Noise simulation of a railway section to be implanted in a densely urbanized neighborhood of São Paulo, Brazil

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Abstract

This article is a case study concerning the operation of a railway line among a dense urban concentration in São Paulo, Brazil. The study involved noise measurements alongside the railway, in order to characterize the train noise emissions and also next to the neighborhood schools (Noise Sensitive Receivers - NSR), distant from the railway axis by about 40m, to characterize the background noise (without train operation). Technical data from the railway track and compositions were gathered as well as the topographical and urban data, in order to compose the digital ground model (DGM) in the noise modelling software. Through the sound measurements, the digital model was adjusted to reproduce the same noise levels as measured at the site (next to the train axis as well as at the neighborhood). Afterwards, noise calculations were performed to determine the sound propagation curves at the neighborhood. Once the train noise impact was analyzed, it were proposed mitigation measures to decrease the noise impact at the nearby schools considering lineside noise barriers next to the railway axis or positioned at the embankment, as well as administrative measures such as varying operational speed and periodicity, in order to attend the Brazilian noise standards. The results showed that mitigating this noise source is complex and depends on the vicinity topography as well as the neighborhood proximity to the railway.

INVITED

Biomedical Acoustics:

Paper ICA2016-649

Holographic tissue engineering using ultrasonic interference patterns

Dolly A. Sanjinez[(a)], Brian D. Patchett[(a)], Natalie C. Sullivan[(a)], Timothy Doyle[(a)]
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Abstract

Biological cells in suspension can be organized into ordered structures by the acoustic forces exerted on them by ultrasonic standing wave fields. To date, only simple tissue patterns such as layers have been constructed using this scaffold-less approach to tissue engineering. This is principally due to the use of single-frequency standing waves transmitted from a single source, producing planar nodes and antinodes in the cell suspension. The purpose of this project was to explore the potential of generating complex tissue microstructures using multiple sources, multiple frequencies, and standing-wave cavities of complex geometry. Multi-frequency sources were constructed by stacking piezoelectric elements of different thickness, and by placing sources at opposite ends of the standing-wave cavity, to generate compound standing waves with complex, non-sinusoidal waveforms. Experimental results with microspheres show that such waveforms can be used to custom tailor the tissue pattern, such as the use of square waves to create thinner cell layers as compared to sine waves, or the use of more complex waveforms to create double-layer structures. Multiple sources were also placed at various angles to each other to generate interference patterns in the standing waves. For example, two sources placed orthogonal to each other generated a square lattice of parallel channels. The experiments demonstrate that standing wave patterns can be produced with levels of complexity higher than simple 2D layers. Computer models also show that the holographic approach should be capable of creating tissue patterns with a 3D complexity similar to that of natural biological structures such as alveoli and lobules. Future research will focus on creating actual tissue structures from these results.

Biomedical Acoustics:

Paper ICA2016-789

Acoustic characterization of enriched and monodisperse ultrasound contrast agents

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Abstract

Monodisperse microbubble ultrasound contrast agents dramatically increase the sensitivity and efficiency in ultrasound imaging and therapy. They can be mass-produced in a microfluidic flowfocusing device at high production rates of up to 1 million bubbles per second. Here, we demonstrate the controlled production of clinically relevant monodisperse bubbles with excellent control over phospholipid monolayer elasticity and microbubble resonance. We also demonstrate the synthesis of bubble suspensions with a narrow size distribution obtained from lab-on-a-chip sorting techniques by pinched flow fractionation or by acoustic bubble sorting. The enriched bubble samples were characterized acoustically using scattering and attenuation measurements for a wide range of pressures and frequencies. Modeled scattering and attenuation coefficients were fitted to the measured data to give the shell parameters for the sorted bubbles. The modeled scattering and attenuation coefficients were obtained by integration of the non-linear response of all bubbles within the acoustic beam of the transmit transducer. Quantitative agreement was found between measured and modeled scattering and attenuation curves for all acoustic characterization pressures using a unique set of shell parameters, both for the overall dynamical response as well as for the absolute magnitude. This confirms that the sorted bubble samples have a uniform acoustic response.

Biomedical Acoustics:

[Paper ICA2016-804]

Laser-driven resonance of light-absorbing ultrasound contrast microbubbles.

Guillaume Lajoinie[(a)], Jeong-yu Lee[(b)], Erik Linnartz[(a)], Joshua Owen[(b)], Pieter Kruizinga[(c)], Nico de Jong[(c)], Gijs Van Soest[(c)], Eleanor Stride[(b)], Michel Versluis[(a)]

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Abstract

The sensitivity of ultrasound imaging is greatly enhanced by the use of microbubble contrast agents through resonant volumetric oscillations. While the increased acoustic contrast is of prime interest for perfusion imaging of organs, microbubbles until now have limited benefit in terms of specificity for ultrasound imaging. Original strategies are required to tackle this difficulty that rely on loading functional targeting ligands onto the microbubble encapsulation. In parallel, another type of wave is used in biomedical imaging that shows great specificity in its interaction with tissue, namely light. This advantage is put to use in photoacoustic imaging where absorbed laser light is converted into a measurable acoustic signal. Here we present a novel ultrasound contrast agent designed to also make use of the superior specificity of laser light. The acoustic agent consists of a gas core encapsulated by an oil layer containing an optically absorbing dye. The resulting laser light absorption can then be used to heat up the gas and drive the system into resonance, thereby generating ultrasound. Combining finite difference simulations and ultra high-speed imaging led to a quantitative physical description of the optical and thermal interactions in the system resulting in the efficient generation of acoustic waves in the MHz range. A range of physical bubble parameters are investigated, in particular those related to the thickness and composition of the light absorbing oil layer. This new generation of contrast agents will open up new applications in medical diagnostic and therapeutic imaging.

Biomedical Acoustics:

[Paper ICA2016-725]

Contralateral suppression of transient evoked otoacoustic emissions in adolescents with and without tinnitus

María Hinalaf [(a, b)], Ana Luz Maggi[(b)], Ester Biassoni[(a)], Mercedes Hüg[(c, d)], Jorge Perez Villalobo[(a)], Karen Grill[(b)], Cecilia Ordoñez[(b)], Andrea Righetti[(e)]

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Abstract

The severity of tinnitus has a high variability; it can potentially cause serious anxiety disorders and in some cases it can even lead to depression. Currently, one of the hypotheses of the genesis of tinnitus involves a deterioration in the functioning of medial olivocochlear system (MOCS). The functioning of the MOCS is evaluated through the contralateral suppression (CS) of transient evoked otoacoustic emissions (TEOAEs) by comparing the amplitudes without and with contralateral acoustic stimulation (CAS). The aim of the present study was to analyze the functioning of the MOCS in adolescents with and without tinnitus through the CS of the TEOAEs. A cross-sectional correlational descriptive study was carried out, involving 77 adolescents (n = 154 ears) with normal hearing with and without tinnitus, who underwent TEOAEs testing without and with CAS using white noise at 50 dB. The results evidenced that the adolescents without tinnitus showed higher global amplitude and higher amplitude in the frequencies 1000, 1500, 2000, and 3000 Hz, in both conditions without and with CAS, in comparison to the adolescents with tinnitus. This

difference was statistically significant ($p<0.05$) in the 1000 Hz frequency and in the global amplitude, without and with CAS. In addition, the adolescents with tinnitus showed less difference between the global amplitudes in the conditions without and with CAS (suppression effect). These results suggest a possible relation between the functioning of the MOCS and the presence or absence of tinnitus, which could contribute to confirm the hypothesis of the involvement of the MOCS in the generation of tinnitus.

Biomedical Acoustics:
Paper ICA2016-364

Discovering new facts and revealing existing myths about the acoustic stethoscope 200 years after its invention

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Abstract
Acoustic stethoscope, invented exactly 200 years ago by French physician, Rene Laennec, is the most widespread medical diagnostic device, and also the icon of the medical profession. Hence, it might seem strange, that the mechanisms and physical phenomena underlying the auscultation examination are still not well understood, and some of the commonly repeated statements – such as those regarding high- or low-pass filtering effects supposedly introduced by specific kinds of chestpieces – have no scientific justification. The present study introduces results of the experimental investigations on the different factors influencing the acoustic properties of a stethoscope. Unlike other studies which were based either on subjective evaluation of sound quality or on measurements of the transfer functions performed using loudspeakers as the primary sound source, the investigations described herein were based on original experimental and data analysis schemes, which allowed to quantitatively evaluate the parameters of signals recorded during the actual auscultation examinations. Results obtained using various kinds of chestpieces and different lengths of the hollow tubes are compared in order to expose the factors which have the greatest impact on the perceived sound quality. It is shown, that the parameters and the construction of the chestpiece determine the achievable sound level, while the hollow tubes act as an acoustic filter. This contradicts the statements repeated in the literature for over 50. years, about the highly frequency-selective behaviour of the diaphragms of the stethoscopes.

Tuesday morning, 6 September 2016 **Dr. Valsecchi Auditorium**
11:00 - 12:00
Architectural acoustics - Room and Building Acoustics
AA5 - Challenges and Solutions in Acoustical Measurements and Design

INVITED
Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-55

Measurements for simulation of speech intelligibility in spaces with conflicting requirements

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Abstract
Computer based measurement and simulation tools are very helpful in architectural acoustics projects. In this paper we will present how these tools were used in several projects of different complexity that presented challenges in aspects of room acoustics and sound system design. Some of the challenges were the limitations of a historical room, a high intelligibility multiple source/multiple receiver sound system design in a room with concave geometry, a meeting space with limited wall area for added absorption and hard surfaces for floor and ceiling, and the conflicting requirements of music and speech intelligibility in rehearsal rooms, among others. All of the spaces had high requirements for speech intelligibility and after measurements and simulations for redesign the goals were met.

INVITED

Challenges and Solutions in Acoustical Measurements and Design:

Paper ICA2016-765

Optimizing floor-ceiling assemblies in wood-framed multifamily buildings using a two-rating method of evaluating impact isolation

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Abstract

In the United States of America (USA), Building Code regulations for airborne and impact sound insulation in multifamily residences are limited and do not extend to all locations and project types. Providing higher levels of sound isolation is driven largely by financial considerations, perceived marketplace and experience. Buildings with higher levels of impact isolation generally have higher construction costs, but also can demand higher rents or selling prices, and developers seek to optimize their designs to balance these constraints. The authors have worked with several large multifamily housing developers to develop separating floor-ceiling assembly designs. This is a cyclical process in which an assembly is designed, tested in the laboratory, built, tested in the field, evaluated with respect to occupant satisfaction; the assembly is then modified as necessary and the process repeats. To evaluate the assemblies in the field and relate to occupant satisfaction, the authors utilize a two-rating method of evaluating impact noise isolation, previously developed [1,2]. The two-rating method has proven more effective than the existing single number ratings for predicting occupant reaction, evaluating products, and providing guidance for future design improvements.

Challenges and Solutions in Acoustical Measurements and Design:

Paper ICA2016-325

A discussion on the uncertainty of absorption characteristics measured by ensemble averaging technique for room acoustics simulations

Toru Otsuru[a], Reiji Tomiku[a], Noriko Okamoto[b], Emi Ueda[c], Asami Nakamura[c]

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Abstract

The authors proposed an in-situ sound absorption measurement method using an ensemble averaging technique, EA metod, for short. Herein, EA method measurements are conducted nine times repeatedly with three pressure-velocity sensors (Microflow; pu-sensor) to prove the effect of sensor difference on resulting sound absorption coefficients and to examine if the uncertainty stays within such a small range as is suitable for room acoustics simulations. EA method con- ducted here follows the configuration given in our earlier papers and sound absorption coefficients of a glass-wool board, a needle felt sheet and a slice form sheet were measured. To prove the effect of pusensor difference on the resulting sound absorption coefficient three pu-sensors were employed. The sensors were calibrated using an acoustic tube with a diameter of 10 cm just before or after a series of measurement. The results of nine times measurement of the glass- wool board showed slight differences only at 100 Hz, 500 Hz and 1250 Hz and uncertainties (standard deviation) of sound absorption coefficient stay less than 0.02. The results of the other two materials are rather in better agreements with less uncertainty. The measurement results of sound absorption coefficient obtained using the three pu-sensors showed good agreements each other for the three materials. The maximum difference 0.04 is observed for the glass-wool board at 1250 Hz. Such larger differences are observed at both limits of the frequency range of the acoustic tube used for pu-sensor calibration. Except the frequencies, uncertainties of sound absorption coefficients measured by EA method with different three pu-sensors stayed the values within suitable range for room acoustics simulations.

INVITED

Calculation Models for Timber Structures (Silent Timber Build):

[Paper ICA2016-132]

Modelling of the tapping machine for finite element prediction tools – Preliminary parametric studies

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Abstract

Dissatisfaction of dwellers apropos acoustic comfort is a common problem often encountered in wooden multi-storey buildings. Such problems could be lessened by addressing vibroacoustic issues during the design phase of buildings if proper prediction tools were available for the engineer. Nevertheless, product development nowadays is still carried out in the aftermath of the construction based on engineers' experience and measurements performed on already existent buildings. The substitution of measurements by easy-to-use numerical predictive models, however, must take place only after those have proven to possess enough accuracy for the predictions carried out. The procedures to evaluate impact sound insulation performance, as described in the ISO 717-2, involve the use of a standardised excitation source: the tapping machine. Even though considerable research concerning the modelling of wooden floor structures (e.g. connections and material properties) has been carried out within recent years, few investigations have been conducted in a way aiming at characterising and modelling the excitation source, its hindering the development of models which could foresee and mimic results stemming from standardised measurements. The study reported on the paper aims at suggesting improvements of low frequency tools for purposes of prognosis by gaining insight about the modelling of the tapping machine. General simplified guidelines for its introduction into finite element commercial software were drawn. Ultimately, a prediction tool combining the knowledge stemming from the investigations performed will enable the determination of the standardised single number ratings obtained from measurements.

Calculation Models for Timber Structures (Silent Timber Build):

[Paper ICA2016-372]

Reverse sea to predict flanking transmission in timber framed constructions

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Abstract

Prediction of Flanking Transmission for lightweight constructions is a sensitive procedure. Lightweight structures are anisotropic and are composed of multiple possibilities of build-ups and structural junctions. The article presents an example of structure composed of Cross Laminated Timber. The modelling strategy to predict such structures is detailed. The framework of the methodology is Statistical Energy Analysis (SEA). SEA involves cutting the structure into subsystems and decomposing the spectrum into third-octaves or octaves. In this way, the exchange of energy flow in the substructures can be analysed. The parameters that govern vibrational transmission between subsystems are damping and coupling loss factors and can be identified experimentally by reversing the problem. The approach is, first, based on sub-structuring strategy of structure; second, on experimentally measured coupling and damping loss factors; third, on extracting vibration level difference to estimate one by one all flanking passes.

Calculation Models for Timber Structures (Silent Timber Build):
[Paper ICA2016-271]

Modelling various floor and wall assemblies and comparisons to measured values

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Abstract
In the project Silent Timber Build a series of calculations have been made in order to find out modelling difficulties using the software SEAWood, which is partly developed and refined in the project. The calculations were carried out in order to achieve a better understanding of which parts of the modelling that is most critical in order to arrive in a satisfactory prediction compared to expected subjective evaluation, hence in order to take measures in the further development of the prediction tool to arrive in better accuracy of the calculations. They are also made in order to find out which building parts that are of less importance or even might be neglected in the assemblies and finally which building parts that are of certain importance in terms of material characteristics and similar, in order to further improve the optimization of wooden floor and wall assemblies. The predicted floor and wall assemblies are typical from Europe (Scandinavia and mid Europe), for which correct and verified laboratory measurements are available in order to perform reliable comparisons and adequate conclusions. The results from the comparisons (predictions vs measurements) show that additional work is needed in order to 1. Refine the model in low frequencies; 2. Focus on improved material characteristics for intermediate layers and also to avoid some of them; 3. Group the floor and wall assemblies due to their prediction ability.

INVITED

Calculation Models for Timber Structures (Silent Timber Build):
[Paper ICA2016-255]

Analysis of acoustical behavior of bare cross laminated timber floors for the evaluation of the improvement of impact sound insulation

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Abstract
The estimation of impact sound insulation of horizontal partitions, evaluated from the performance of basic components using EN 12354-2 Standard, do not actually provides satisfactory results when applied to the floors realized with cross laminated timber (CLT) elements. Among the possible reasons of this limited correspondence between predicted and measured impact noise values, one of the most interesting is the difficult to correlate the reduction of the impact sound pressure level of the floor covering, measured in laboratory on a concrete slab, with the actual behavior on a bare CLT floor. In this paper the results of a laboratory evaluations independently managed by different researchers on similar CLT structures is reported. The purpose of this study is to identify an empirical spectrum of the normalized impact sound pressure level of a reference floor realized based on CLT technology, in order to provide an useful and simple tool for estimate the noise insulation performances for this type of building element.

Calculation Models for Timber Structures (Silent Timber Build):
Paper ICA2016-159

Hybrid cross-laminated timber floors. Comparison of measurements and calculations

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Abstract
During the last years, there has been an increasing interest of cross-laminated timber (CLT) constructions among project owners, architects and producers. A more extensive use of wood in buildings is also of strategic interest in the wood industry. Design solutions to fulfil sound insulation requirements between apartments have been an issue on earlier work beside research work on flanking transmission with CLT elements. Recently we recognize an increased interest on CLT solutions used in other building categories, for instance student apartments and schools. Development and verification of floor constructions is of course important also for such applications. The paper will present a collection of results from SINTEF Building & Infrastructure combined with some preliminary research work running in the "Silent Timber Build" project within the WoodWisdom-Net program. The paper will focus on impact sound insulation properties of hybrid solutions with CLT and concrete. Results from both laboratory and field measurements will be given. In addition, calculations based on analytical methods have been carried out. A preliminary comparison between calculation and measurement results of the impact sound insulation will therefore be given in the paper. It is possible to fulfil the main impact sound requirement in Norway based on optimization of such hybrid solutions.

Tuesday morning, 6 September 2016 **Cardenal Pironio Auditorium**
11:00 - 12:00
Architectural acoustics - Room and Building Acoustics
AA4 - Calculation Models for Timber Structures (Silent Timber Build)

Calculation Models for Timber Structures (Silent Timber Build):
Paper ICA2016-163

SEAP – Acoustic design tool for Stora Enso Building Solutions

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Abstract
A new web based acoustic design tool is developed for wooden building systems from Stora Enso being a leading provider of renewable solutions in packaging, biomaterials, wood and paper on global markets. The first edition covering not only acoustics but also static design is now launched on https://engineer.clt.info/. The first version of the acoustic calculation tool SEAP (Stora Enso Acoustic Prediction tool) covers various floor assemblies and wall assemblies and will be available during 2016. The frequency range considered is 50 Hz – 5000 Hz, hence adaptation terms $C_{I,50\text{-}2500}$, $C_{50\text{-}3150}$ and $C_{50\text{-}5000}$ might be calculated and evaluated in order to secure low frequency structural design, covering the important frequency range from 50 Hz according to ISO 717. This is necessary in order to follow recommendations regarding future requirements proposed in the final report from the COST action TU 0901. Introducing a prediction tool will facilitate users to predict sound insulation in wooden structures using building system by Stora Enso in the design stage. The framework and layout of the design tool is developed by Stora Enso and the mathematical algorithms are developed by experienced acoustic researchers from WSP, Akustikverkstan and Lund University. The model is based on a modular system adding e.g. various floor packages and ceiling packages to the CLT structural system by Stora Enso. The model build up is made in order to perform fast and efficient updates as soon as new products are implemented or material characteristics are changed. The model is carefully evaluated piece by piece in order to fulfill high requirements for accuracy.

Calculation Models for Timber Structures (Silent Timber Build):
Paper ICA2016-165

Measurement series to verify the accuracy of Stora Enso Acoustic Prediction tool - SEAP

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Abstract
Stora Enso Acoustic Prediction tool (SEAP) is developed in order to secure a high level of accuracy in the design stage. During development a number of previously performed laboratory measurement were used in order to design the model and secure the calculated results. A third party evaluation of the draft model raised some doubts regarding the model accuracy and after some investigations it was clear that some of the laboratory measurements suffered from shortcomings and contained results that were difficult to explain, hence also creating deviations in the model. This applied to both impact sound level and airborne sound insulation. However, the model itself is designed in order to always make it possible to introduce new materials in the model as they enter the market but also to improve the accuracy as soon as new knowledge is available. Due to the uncertainties raised and in order to have the best basis prior to introduce the first version of SEAP it was decided to carry out a carefully designed laboratory measurement series and from that adjust the model to secure that the calculated results will fall within the accuracy requirements. The results from the measurement series show that it was necessary to realize some adjustments in order to reach the high requirements of the model accuracy. From the updated model new calculations were made and now the model accuracy is satisfactory. All calculations performed on various floor assemblies now fall within the accuracy limits for this stage of the model development. The model will now be further developed covering the delivery from Stora Enso Building Solutions, i.e. to facilitate calculations for a complete building.

Calculation Models for Timber Structures (Silent Timber Build):
Paper ICA2016-683

Effect of difference in specification on sound insulation in cross laminated timber separation wall

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Abstract
The Act for Promotion of Use of Wood in Public Buildings (Law No. 36 of 2010 of Japan) was enforced in 2010. The promotion of the use of wood can contribute to the prevention of global warming, etc. As this act, a public building in a low layer is supposed to attempt making to timber construction. Moreover, CLT (Cross Laminated Timber) was standardized by JAS (Japanese Agricultural Standard) in 2014. CLT has been used in Europe, especially in Austria. However, we have little knowledge about CLT's performance; structure performance, fire protection performance and sound insulation, etc. Among the objections and the troubles of houses in Japan, the sound insulation is one of the most serious issues. Besides, a sound insulation of the timber construction is low compared with that of the concrete construction. It is necessary that the sound insulation performance is more than the sound reduction index Rr-40 as separation wall between each unit of apartment houses by the Building Standard Law of Japan. Therefore, we have measured the sound insulation of CLT separation wall in the reverberation chambers. The 150 mm thickness CLT, which had 5-ply and 5-layer, was basics and the specimens were changed the specification of extra gypsum board wall for one side and both side, extra gypsum board wall with independent furring strip from CLT panel. In this paper, the effect of difference in specification on sound insulation in CLT separation wall was investigated. The specimen, which had the specification of extra gypsum board wall for one side with independent furring strip from CLT panel, was shown to be effective to improve the sound insulation of CLT separation wall.

Legislation and Regulations in Building Acoustics:
Paper ICA2016-180

Establishment of indoor noise regulation between adjacency households in apartment building

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Abstract
Since Apartments share walls and floors by many households, various noises may be transferred to neighboring households. In particular, floor impact sound generated by vibration of floor plates is the most offensive noise. If a performance of vibration reduction in the floor in apartments is excellent, floor impact sound transferred to the below household will also be reduced. However, a level of noise transfered to the neighboring households may vary even in the same performance apartments due to excessive behavior of residents. That is, a noise generation behavior of residents can become an important factor that determines an amount of noise transfer. This study aimed to set up criteria of impact noise during daily living to reduce the excessive impact noise generated behavior by residents. Its reference value was presented as 43 dBAthrough performance evaluation on floor structure and psycho-acoustic experiment.

Legislation and Regulations in Building Acoustics:
Paper ICA2016-301

Suggest toilet noise standard for multi-dwelling residential building

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Abstract
Toilets are items that are pointed out most frequently for noise among multi-dwelling residential building facility equipment. It is difficult to adjust the time of usage due to the nature of the toilet, which may interfere with the user's sleep if it is adjacent to the main room. The toilet pipes of most Korean multi-dwelling residential buildings are vulnerable to noise from generating airborne sounds generated by pipes as an under-slab plumbing structure passing through the ceiling of lower households by penetrating the floor slab. This study has proposed a toilet noise standard in order to provide a calm environment for residents to induce the development of technology and construction methods for reducing toilet noise. Present conditions have been identified in 30 households with under-slab plumbing and 30 households with on-slab plumbing according to the wastewater plumbing structure to evaluate the degree of annoyance while draining the toilet bowl. As a result, a noise standard of 45dBAhas been proposed when the adjacent household uses the toilet. The mean value of annoyance during the numeric scale evaluation was 4.9 while the ratio of response during the verbal scale evaluation (maximum noise level of 46dB(A)) has been indicated as showing ‘slightly’ 3.51%, ‘moderately’ 28.1%, ‘very’ 36.8%, and ‘extremely’ 31.6%. Also, the ratio of households satisfying the standard when performed on new multi-dwelling residential buildings is roughly 40%.

INVITED

Legislation and Regulations in Building Acoustics:

Paper ICA2016-487

Comparison of acoustic regulations for housing and schools in selected countries in Europe and South America – A pilot study

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Abstract

Acoustic regulations for housing and schools exist in most countries in Europe, the main reasons being protection of health of citizens in their homes and optimizing learning and work conditions in schools. Comparative studies in Europe have shown a high diversity of descriptors and limit values for acoustic requirements. Considering globalization and noise as a health issue, it is important also to extend attention to other parts of the world and establish dialogue and hopefully cooperation, thus facilitating exchange of experience with construction solutions fulfilling different levels of requirements. As a pilot study, acoustic regulations in three countries in South America, namely Argentina, Brazil and Chile, have been considered. The findings indicate weaker requirements than typical in Europe, and at both continents there is a joint challenge to review regulatory requirements in those countries with a high need for improvement of acoustic conditions in housing and schools. It is concluded that one of the first steps is to create awareness among authorities and building industry and to exchange experience about construction solutions, which in many cases are quite simple. The paper includes examples of specific acoustic requirements on airborne and impact sound insulation, noise from traffic and from service equipment for housing and schools and in addition on reverberation time for class rooms and provides the basis for discussing future cooperation on optimizing acoustic regulations.

Legislation and Regulations in Building Acoustics:

Paper ICA2016-76

Establishment of the acoustical standard of the Korean classrooms using speech intelligibility test

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Abstract

The most important function of the classroom is to transmit educational information from teachers to students more accurately and clearly. Acoustical environment of the classroom, thus, has an important effect on the improvement of students' learning ability. To provide appropriate acoustical environment for learning to students, it is necessary to make acoustical performance criteria for classrooms and to make a guideline for designing classrooms. However, in Korea, there has not been the acoustical standard for classrooms, so it is difficult to control and manage appropriate acoustical performance when designing and building classrooms. The present study aims to suggest acoustic performance criteria for classrooms which are suitable for the Korea language. In order to this, standard classrooms were made by standardizing architectural dimension of 17 middle and high school classrooms in Cheongju. The speech intelligibility tests were carried out using three different languages including Korean, English and Chinese. 20 native speakers for each language were used as subjects for the speech intelligibility tests.Finally, auralized sound source was made under 5-step conditions of reverberation time (0.47 sec ~ 1.22 sec) by changing indoor sound absorption of a real classroom. Listening tests were undertaken to 52 Korean adults with normal hearing, using the auralized sound source. The results proved that the most appropriate reverberation time for learning was above 0.76 sec. Based on the research findings, ideal acoustical performance criteria for the classrooms in Korea as follows: background noise is below 35dB(A), and reverberation time is below 0.80 sec. Also, it is necessary that indoor sound absorption should be above 20% without sound absorption on side walls in order to satisfy with the acoustical performance criteria.

Legislation and Regulations in Building Acoustics:
Paper ICA2016-716

The new standard DIN 18041 – acoustic quality in rooms

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Abstract
The since 1968 well established DIN18041 standard was revised from October 2013 to mid 2015 to commit the room acoustic requirements for the implementation of the inclusion in the field of hearing and to take into account trends in modern architecture. In addition to these technical and social aspects DIN 18041 with the new title “Acoustic quality in rooms – requirements, recommendations and instructions for planning” of 2016 gives clarifications and additions as well as deletions compared to the 2004 edition. The revision of DIN 18041 provides clear and unambiguous guidelines described as requirements and recommendations for everyday rooms where the mutual listening and understanding but also finding quietness is significantly important.

Tuesday morning, 6 September 2016 **Room 204**
11:00 - 12:00
Architectural acoustics - Room and Building Acoustics
AA8 - Legislation and Regulations in Building Acoustics

INVITED
Legislation and Regulations in Building Acoustics:
Paper ICA2016-811

Acoustics evolution of official constructive solutions for impact noise in the period 2005-2014 in Chile

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Abstract
Following earlier work focused on airborne noise insulation, this study aimed to analyze the evolution of the official list of constructive solutions during the period 2004-2015, which makes it mandatory acoustic insulation between dwellings. This regulatory body include in-situ and laboratory tests to be included in an official list managed by the Ministry of Housing. This communication compares the main descriptive parameters of normalized impact sound pressure level in the period 2005-2014 and its implications for the changes the country has had in the decade considered.

INVITED
Legislation and Regulations in Building Acoustics:
Paper ICA2016-160

Translation of existing impact sound insulation descriptors into new proposed ones, based on a large set of in situ measurements

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Abstract
The need for revision and harmonization of sound insulation descriptors is generally accepted among most building acoustics specialists. Undoubtedly, if all countries used the same descriptors, it would be a great advantage to all the sectors related to the building construction industry, legislators and final users. Concerning sound insulation regulations, the most common impact sound insulation

descriptors used are L'nT,w and L'n,w. In some countries, the spectral adaptation term CI is also considered, including different frequency ranges, which widens the choices for impact sound insulation descriptors used in regulations. The purpose of this paper is to determine, based on a large set of in situ measurements, how existing impact sound insulation descriptors relate to new proposed ones, in order to be able to make empirical translations of descriptors. The effect of the spectral adaptation term CI and the effect of the building system (heavy/light floors) are also analysed. Based on the translation equations found, one of the main conclusions is that heavy and light floors yield different empirical translation equations and that it would not be correct to use the same translation equation for all types of floors.

INVITED

Legislation and Regulations in Building Acoustics:

Paper ICA2016-861

Requirements and building guidelines for the retrofitting of traditional timber floors

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Abstract
In Belgium, a large part of urban dwellings consist of 19th century buildings. Under well defined circumstances, the existing standard allows to deviate from the acoustic requirements for these old buildings. However, for a future revision the proposal to limit this deviation and to impose a minimal requirement is examined. Improving existing timber floors in buildings is not an easy task since many technical and space-related restraints usually apply. Often only adaptations above or under the load-bearing joists are possible, sometimes both. In a few experimental studies, solutions are sought within these restrictions, focusing on impact sound insulation. The measurements allowed to gain insight in different transmission mechanisms. The results reveal the possibilities but also the limitations of several improvement techniques. Finally, an adapted requirement is proposed.

Tuesday morning, 6 September 2016 **Auditorium 2**
09:40 - 10:40
Numerical Techniques
NT2 - Numerical Techniques (others)

Numerical Techniques (others):

Paper ICA2016-336

On acoustics of diesel particulate filters

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Abstract
With the increasing share of diesel engines in the vehicle market, air pollution and environmental impacts such as noise began to be the subject of examination. Using diesel particulate filter (DPF) on cars is becoming a standard in many countries in order to reduce the ill effects of diesel exhaust gases which have a significant role in air pollution. Although the main purpose of DPFs is to reduce harmful emission of the soot particles, they also effect acoustic emissions. In this paper, a finite element analysis is used to model the full DPF to predict acoustic performance of it. Transmission Loss is chosen as the parameter that defines the acoustic performance of DPF. Based on the FEM computation, the transmission losses under the idle condition of a diesel engine are calculated and the frequency-dependent sound transmission loss was plotted in the entire plane wave range.

Numerical Techniques (others):

Paper ICA2016-362

Predictions of the far-field noise due to vortex-shedding using a Lattice Boltzmann scheme and an acoustic analogy

José Pedro de Santana Neto[a], Danilo Braga[a], Andrey Ricardo Da Silva[a], Julio Cordioli[a]

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Abstract

This work investigates the generation and propagation of tonal noise due to vortex shedding at low Reynolds and Mach numbers. The investigations are conducted with a simple numerical scheme based on the Lattice Boltzmann method and the acoustic analogy based on the twodimensional formulation of the Ffowcs Williams and Hawkings equation. The simulations are conducted by using a squared cross section for a fixed Mach number (M=0.2). The predictions of the far-field noise are compared with the data available in the literature for the same geometric conditions. In general, the results confirm the dominance of the dipole sound generation at a narrow frequency band for the flow parameters considered in this investigation. The good agreement with the literature results reinforce the fact that the Lattice Boltzmann can capture reasonably well the sound generation phenomena associated with vortex shedding of laminar boundary layers.

Numerical Techniques (others):

Paper ICA2016-371

Room acoustics design using SWARM optimization method

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Abstract

This work presents an acoustic inverse problem, where the absorption coefficients are obtained from the Impulse Response (IR) and from the knowledge of the room geometry, source and receive locations.The parameter extraction is achieved by the particle swarm optimization (PSO) method, where a given impulse response —measured in a real room or with some desired characteristics— is compared with simulated ones. The IR simulation is performed by an improved ray-tracing method, which clusters the first reflections. Several issues are addressed in the paper, such as the reflection order influence and the multiple solutions according to the objective function. The paper describes how to combine the Ray-tracing and the Swarm methods in order to extract the absorption coefficients. It is also presented the optimization algorithm performance for several objective functions, such as impulse response, decay curves, magnitude spectrum and other error criteria. The convergerate and final parameter error are also investigated in order to present the applicability of the proposed method and to define the most apropriate objetive function to solve the inverse problem.

Numerical Techniques (others):
Paper ICA2016-523

Multiple-scattering theory for two-dimensional arbitrarily shaped acoustic composites

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Abstract
Multi-scattering theory (MST) is an analytical tool which allows for the resolution of the field dispersed (scattered) by an interfering medium after leaving a source. The basis of the method consists in determining the way each inhomogeneity (scatterer) in the medium affects the incident wave, something which ultimately depends on the shape and mechanical properties of the scatterer. Unlike previously reported MST implementations in acoustics, which only deal with cylindrical scatterers, we studied the effects of employing structures with different shapes (triangular, square, elliptical). The using of these scatterers to conform periodic composite materials (crystals) is further tested and the arising results are contrasted with predictions based upon the band structures of the corresponding crystals. Also, both the geometrical parameters of the scatterers (cross-section and size) and their orientations in the crystal with respect to the source are varied, giving rise to additional tools to manipulate the acoustic field. All field computations were performed using a GPU.

Numerical Techniques (others):
Paper ICA2016-118

Use of boundary conditions of impedance, continuity, Sommerfeld, for evaluating sound attenuation of mixed passive silencers with FEM-2D

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Abstract
In this paper we study the influence and importance of boundary conditions (BC) in the formulation, implementation, and solution for acoustic problem silencers in mixed passive silencers. The analyzed model corresponds to a muffler chamber expansion, using FEM-2D (two-dimensional Finite Element Method). As a first approximation to describe the dissipative condition of chamber walls, we define an impedance BC; but it forces a certain idealized acoustic behavior of the muffler. In order to achieve more reliable and accurate predictions, we model the impedance walls as an absorbent sub-domain, which is coupled with the rest part of the muffler. Due to the characteristics of the problem (knowledge of the physical phenomena) was necessary to fix conditions like Impedance, Continuity and Sommerfeld. Delany-Bazley approximation is used to describe the property of porosity of the absorbent material and medium. The introduction of the absorbent sub-domain makes the sound pressure at each point of the domain more sensitive ("real") in this kind of muffler and yields to better results. The calculated transmission loss values show good agreement with the experimental measurements.

Numerical Techniques (others):

[Paper ICA2016-133](#)

Muffler shape optimization accounting for acoustic-structure interaction

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Abstract

A computational model for muffler shape optimization using genetic algorithms is presented in this paper. The objective is to minimize the shell and outlet sound radiation of different mufflers as well as to maximize their transmission loss in a given frequency band. The acoustic-structure interaction modeling is performed using the finite element method for the interior and exterior problems, and accounts for axial symmetry. Perfectly matched layers are considered in the exterior domain to simulate free-field radiation condition. Furthermore, the effect of absorptive material in the muffler is included using the Allard-Champoux model. Finally, the pressure drop and feasible dimensions are used as constraints for the design. Results for sound transmission loss and directivity for different types of optimized mufflers are presented.

Tuesday morning, 6 September 2016 — **Auditorium 3**

09:20 - 10:40

Physical Acoustics

PA2 - Sonochemistry and Sonoprocessing

INVITED

Sonochemistry and Sonoprocessing:

[Paper ICA2016-20](#)

The use of ultrasound to enhance crystallization of minerals from concentrated saline effluent

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Abstract

Many effluent treatment processes result in a concentrated brine solution that requires disposal. Bulk crystallisation of the salt can facilitate this disposal, by providing a solid that can be more readily transported. In this presentation, we consider the use of ultrasound to assist in initiating the crystallisation of salt from saline effluent sourced from the Australian dairy industry. The effluent has already been concentrated to 22 or 30 wt% total solids, these being principally sodium chloride, but with some calcium phosphate and lactose. The precipitation follows reverse solubility behaviour with respect to temperature, with the shortest induction times at the highest temperature. This reflects the low solubility of the calcium phosphate salts, which precipitate first. Ultrasound can significantly reduce the time required to induce crystallisation. The use of a higher power produces a greater reduction in crystallisation time, relative to a lower power at the same energy density. Further, the use of two pulses of ultrasound during the induction period, also shortens this induction period, relative to the same energy application as a single pulse. Indeed, using a double pulse reduces the induction time to around 30 minutes, relative to 2.6 hours without ultrasound. Ultrasound also influences the crystal morphology, with needle like crystals only appearing when this is utilised. However, the ultrasound cannot improve the crystal yield, which is limited by the relative humidity of the atmosphere above the crystallising solution.

INVITED

Sonochemistry and Sonoprocessing:

Paper ICA2016-202

Understanding and optimising sonication conditions for crystallisation processes

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Abstract

Crystallisation is a common technique adopted by many industries for separation and purification operations but it can suffer from unpredictable nucleation rates and inconsistencies in product qualities. To remediate these problems, ultrasound has been used to facilitate and control nucleation of crystals (sonocrystallisation), resulting in a more consistent and narrower size distribution of product. However, the mechanism behind sonocrystallization process is contentious and often reported is the continuous sonication of the system, which causes issues such as heating, probe surface erosion and cost of additional input energy. This study examines the influence of cavitation activity and different modes of pulse sonication on the crystallisation process. The results show a close association between cavitation activity, quantified in terms of sonoluminescence intensity, and the minimum crystal size obtained. In addition, the initial short sonication time 5 seconds proves to be effective for sonocrystallisation, showing results that are comparable to systems undergone 90seconds of sonication.

INVITED

Sonochemistry and Sonoprocessing:

Paper ICA2016-59

Study about the application of acoustic agglomeration of aerosols for source term mitigation in nuclear accidents

Manuel Aleixandre[a], Enrique Riera[a], Rosario Delgado-Tardáguila[b], Luis E. Herranz[b], Juan A. Gallego-Juárez[a]

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Abstract

During severe accidents of nuclear power plants radioactive material can be released from fuel and form aerosols that, even though highly unlikely, might reach the environment. There exist a number of systems to mitigate any potential emission from a nuclear power plant and, in particular, what is known as Filtered Containment Venting System (FCVS). After the Fukushima accident some investigations have been launched to boost the efficiency of those systems as much as possible under any foreseen conditions. This work deals with an experimental study about the potential application of a power ultrasonic system as an innovative approach to precondition the particle load reaching the first filtration stages of the FCVS. This study includes the design and acoustic characterization of an ultrasonic agglomeration chamber at 21 kHz in which a high intensity standing wave field is established. This system has been tested with SiO_2 aerosols of diameters of 0.3 µm, 1 µm and 2.5 µm, and polydisperse TiO_2 aerosols. The agglomeration effects have been studied, and the results compared with the theoretical model predictions.

INVITED

Sonochemistry and Sonoprocessing:

Paper ICA2016-107

Acoustic design aspects of megasonic reactors for oils and fat separation

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[c] Karlsruhe Institute of Technology, Germany

Abstract

The performance of ultrasound reactors developed to enhance oil and fat droplet separation and fractionation rely on a number of factors. These factors include the selection of transducers, the reactor geometry and dimensions, and the physical properties of the product to be processed. Ultrasound standing wave separation reactors operating in the MHz frequency range are also referred to as megasonic reactors. The separation principle is based on acoustic radiation forces following Gorkov's theory. Depending on density and compressibility, fluid droplets are splitting between nodes and antinodes in a standing wave field. The interactions between primary and secondary acoustic radiation forces and acoustic streaming determine the reactor's performance. The efficiency of the megasonic reactor strongly depends on the attenuation of the sound waves by the fluid. The sound pressure levels obtained in a large scale (up to 1.2 m) trough attached with a 600 kHz transducer were measured in water or oil bearing materials using a novel hydrophone system. Significant attenuation was observed from the midpoint of the trough through to the end when sound transmission was investigated in water, while further attenuation was observed when using palm oil sludge. Following these results, a methodology to validate megasonically enhanced oil separation from avocado and palm oil at minimum sound penetration levels at laboratory scale was developed. The combination of sound penetration data in sludges or paste supported by benchtop results will provide the information to design large scale megasonic reactors.

Tuesday morning, 6 September 2016 **Auditorium 3**

11:00 - 12:00

Physical Acoustics

PA2 - Sonochemistry and Sonoprocessing

Sonochemistry and Sonoprocessing:

Paper ICA2016-374

High frequency sonoluminescence in sulphuric acid and phosphoric acid

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Abstract

Multibubblesonoluminescence (MBSL) emission spectra at different forcing frequencies (20, 580, 850 KHz and 1.1 MHz) in Sulphuric acid and Phosphoric acid, under different noble gases atmosphere (Ar, Xe and He) have been investigated. Using lower frequency we observed the presence of new emissions lines which may indicate the formation of ionized species with very high energy level inside the bubbles. Since it was recently demonstrated that MBSL at higher frequencies evidences the formation of line emission from OH radicals due to non-thermal mechanism, supporting also the calculation which suggest that highest temperature can be achieved at the bubbles core, we subjected these inorganic acids, saturated with noble gases, to high power, high frequency and also focused ultrasonic fields. The emission spectra revealed that the presence of emission lines is strongly reduced, probably due to the more symmetric bubbles collapse at these frequencies, due also to higher viscosity of these liquids respect to the water. On the other hand the contribution in the far UV region confirms the presence of ionized species due non thermal plasma formation although no significant differences in the emission spectra between focused and non-focused ultrasonic fields have been observed.

Sonochemistry and Sonoprocessing:

Paper ICA2016-90

New ultrasonic technologies for food dehydration process intensification

Roque R. Andrés[(a)], Alfonso Blanco[(a)], Enrique Riera[(a)], Ángel Guinot[(a)]
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Abstract

Food dehydration processes assisted by power ultrasound constitute a new, efficient and green technology. In order to obtain good results with this technology it is necessary to take into account several aspects regarding the ultrasonic generation, the energy propagation and absorption in the samples. Ultrasonic waves produce different effects when propagating through a medium, like an increase in mass transport kinetics by accelerating these kind of processes, and others related to the so called sponge effect and cavitation. This kind of processes needs the whole system to work in a high-power regime. This may imply the appearance of non-linear effects in the transducer behaviour and in the acoustic field generated inside the dehydration chamber. The aim of this work is to describe this technology, capable of producing permanent changes in the samples treated, considering the design and characterization of high-power ultrasonic transducers, the generation and performance of the high-intensity ultrasonic field and the ultrasonic effects produced in the samples, and paying especial attention to those non-linear effects that may appear and its influence on the global behaviour of the whole system.

Sonochemistry and Sonoprocessing:

Paper ICA2016-580

Magnification of atomic lines resolution and optics of bubbles originated by laser breakdown of salt water in ultrasound field

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Abstract

Research studies have been carried out the acoustic effects accompanying optical breakdown in water generated by focused laser and ultrasound radiation. It is shown, that the method of registration of acoustic emission from a breakdown zone allows to investigate thresholds and dynamics of laser breakdown which will be in accord with high-speed optical methods. For the first time the experimental results are obtained, that show the sharply strengthens effects of acoustic emission from a breakdown zone by the joint influence of a laser and ultrasonic irradiation. Experiments were performed by using 532 nm pulses from a Brilliant B Nd:YAG laser. Acoustic radiation was produced by acoustic focusing systems in the form of hemispheres and rings at various frequencies. Essentially various thresholds of breakdown and character of acoustic emission in fresh and sea water are found out. For the first time the experimental result is established, testifying that acoustic emission of optical breakdown of sea water at presence and at absence of ultrasound essentially exceeds acoustic emission in fresh water. Atomic lines of some chemical elements like sodium and calcium were investigated for laser breakdown of water with the ultrasound field. The effect of magnification of these atomic lines resolution for salt water in ultrasound field was obtained. It is shown that the method of registration of acoustic emission from a breakdown zone allows to investigate thresholds and dynamics of laser breakdown which will be in accord with high-speed optical methods. The study revealed important practical applications of acoustic emission for breakdown and diagnostics of cavitation in opaque environments.

POSTER

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-57

Objective detection method of sound coloration in electroacoustic enhancement system

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Abstract
The Active Field Control (AFC) is an electroacoustic enhancement system to improve the acoustic conditions of a space. In this system, how to prevent sound coloration caused by acoustic feedback is the key procedure in a tuning process. In order to incorporate the control of coloration into the automation of tuning, an objective detection method is studied based on subjective assessment and the objective evaluation of coloration. In order to objectively evaluate the coloration, two responses are calculated from the impulse response (IR) captured in the field. One is a frequency characteristic of the Decay-Cancelled IR. The other is a moving-average frequency characteristic based on the previous response. The index parameter σ_G is calculated by the standard deviation of the ratio of these responses. Subjective assessment was carried out using IRs captured in several sound fields with AFC and it is found that σ_G can be an objective evaluation index of coloration. AFC consists of four independent acoustic feedback channels. If the peaks of the frequency characteristics in each feedback channel are suppressed below a certain level, it is considered that subjective detection of coloration might be minimized. The detection and adjustment of frequencies perceived as coloration is carried out in several steps: i) IR is measured at each open loop channel of the system; ii) Several peaks in the frequency characteristics of the late part of the Decay-Cancelled IR are selected since generally coloration is detected in the late part of the IR; iii) These peaks are adjusted at the averages of the maximum values of the frequency characteristics. As a result, subjective coloration is found to be prevented in the AFC auto tuning process with σ_G.

POSTER

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-442

The acoustic performance of walls made with sustainable and industrialized panels

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Abstract
The aim of this paper is the evaluation of the acoustic performance of a modular construction system. Different kind of materials and construction systems, which are focused in the reduction of wastes, like the modular system, are increasing in the Market. The development of materials and construction systems which integrate residues in its composition is a growing tendency in the Brazilian building sector. There are, nowadays, sustainable and industrialized panels for sealing walls, such as Oriented Strand Boards (OSB), Cement-bonded wood particle board (CBWP), Fiber reinforced cement boards and gypsum plasterboard. These panels have found their place in the sealing walls market because of their better performance and lower costs, when compared to conventional sealing materials, such as concrete or ceramic masonries. This paper shows the acoustic performance of a modular construction

system, which is built with Steel framing and the different kinds of panels described above. The measurements were performed according to both ISO 16283, and ISO 10052. Results show that the acoustic performance of the modular construction system, measured according to ISO 16283 parameters, is better than those from traditional walls built with concrete or ceramic masonries, in regard to sound insulation. Uncertainties in the simplified method (by ISO 10052) are also pointed out. The acoustic performance of the modular construction system, according the ISO 16283 parameters, showed that they have a better performance than those which are built with ceramic or concrete masonries. Also, the errors in the simplified method (by ISO 10052) are found. They were classified accordingly with the Brazilian law.

Tuesday morning, 6 September 2016 **Lounge Lateral Room**
10:20 - 10:40
POSTER SESSION - Monitor 1
Architectural Acoustics - Room and Building Acoustics
AA3 - Architectural Acoustics For Non-Performance Spaces

POSTER

Architectural Acoustics for Non-Performance Spaces:

Paper ICA2016-409

Ultra low frequency floating floors controling resonances

Alexandre Boratto
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Abstract

Acoustics interactions of mechanical and dimensional order have the capacity of modify sound environment response. We suggests possible point of work when is a planning of projecting sound isolated studio or a control room. Multiple variations could happen in relation to acoustic frequencies reinforcement and attenuation. This depends on properties like geometry of existing and active elements, including size, shape, thickness and associated resonances. Usually rooms have irregularities below 300 Hz, due to acoustic modes, phase, mean free path, and resonances. In order to provide a better acoustic response, we describe the technique of ultra low frequency floating floors. Floors almost are the largest available surfaces in concern the low’s frequencies control. The proper acoustic coupling required using elements with value in terms of that spectrum with longer wavelength, and that should not enter more active resonances in the acoustic environment. Each module resonance, in this technique, could be calibrated to below 20 Hz, as well to projecting division and positioning of the floor’s plane in parts with assignments for different resonance frequency attenuation. The energy consumption of large surfaces with very low resonance also attenuates electively correspondent’s upper harmonics. Also could be observed less damage at frequencies between 35-370 Hz using this frequency correction process, We’ve usually search, on data from studios, related problems of irregularities in bass response, associated with various panel and bass traps commonly used. Floating floors has also acoustic insulation properties, resulting in more benefits due to cost. In the last 12 years, our team has successful projected and tested control rooms and studios with that technique.

POSTER

Architectural Acoustics for Non-Performance Spaces:

Paper ICA2016-707

Acoustic projects of contemporaries open plan offices in Brazil

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Abstract

The main factor responsible for worker satisfaction is his workplace. An adequate acoustic treatment in those places is essential for employee's satisfaction. Once the noise is the main source of complaint in open plan offices, it is fundamental that the project includes acoustical solutions that control sound levels, such as the using of highly absorbent ceiling tiles and absorbent screen partitions with an adequate height between workstations. However, as the reduction of noise levels can cause loss of speech privacy in the workplace, the use of sound masking systems is recommend in order to mask speech sounds and eliminate unwanted sound frequencies. Although the use of this set of guidelines is fundamental for the development of a good project, this article shows, through research with the most important acoustical consultants in Brazil, the reasons why only part of these solutions are applied in this country.

POSTER

Architectural Acoustics for Non-Performance Spaces:

Paper ICA2016-704

Acoustic insulation facade of historical building that will be transformed in the first Brazilian 6 stars hotel

José Carlos Giner[a], Raquel Rossatto Rocha[b]

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Abstract

The project consists in restoring the old maternity Filomena Matarazzo located in the heart of São Paulo - Brazil and transform the old construction in the first 6 stars hotel in Brazil, a real challenge in acoustic terms. The building is listed by the Brazilian Historical Buildings (HPB), a true landmark for the city. The architectural project is by architect Jean Nouvel and designer Philippe Stark. After getting acquaintance with the different requirements of comfort the suites, that included maximum sound pressure level inside, the project goal was to evaluate and determine the appropriate acoustic insulation of the facade of the building in order to meet the required criteria, due to the intense noise generated by air and road transport. To accomplish that, first we made local soundscape measurements for computer modeling of noise map CadnaA software. The next step will be simulation in software SONarchitect the performance of acoustic insulation required to meet the owners requirements. In addition, it is intended compare the results of sound insulation in loco of the composite facade, with measurements of Weighted Standardized Level Difference at 2 meters from façade (D2m,nT,w) the current facade mainly focused on the windows, before and during construction, so we can verify the data of computational modeling with measurement data.

POSTER

Architectural Acoustics for Non-Performance Spaces:

Paper ICA2016-162

Acoustics and architecture in office buildings: how the site plan and the shape of the building affect the levels of incident noise in facades

Andrea Destefani[a], Maria Akutsu[b], Marcelo de Mello Aquilino[c], Cristina Y. Kawakita Ikeda[d]

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Abstract

The absence of any soundscape analysis in the early stages of the building construction process is the norm in real estate developments in Brazil. Acoustical consulting only begins after the location, the site plan and the shape of the building have already been defined, limiting the possibilities of solutions that may reduce the acoustic demands for facades. With the increase in noise pollution in Brazilian urban areas, it is important to emphasize that the acoustic consulting should be initiated during the land prospecting phase in order to improve the possibilities of achieving a more effective acoustical performance of the building. This article aims to demonstrate the extent of sound mapping as a tool for decision-making, whether in the site plan studies or in the definition of the shape of the corporate commercial buildings (Class AAA) located in disadvantaged acoustical environments due to intense levels of traffic noise. Therefore, this paper presents a case study in which computer noise simulations were carried out considering the incident noise on a real building, with two alternative proposals for the architectural design which enable lower values of noise incident on the facades to be achieved.

Tuesday morning, 6 September 2016 **Lounge Lateral Room**
09:40 - 10:00
POSTER SESSION - Monitor 2
Noise: Sources and Control
NS1 - Aircraft Noise – Aeroacoustics

POSTER

Aircraft Noise - Aeroacoustics:

Paper ICA2016-615

Validation of cold jet noise rig at Laboratory of Acoustics and Vibration (LVA), Federal University of Santa Catarina (UFSC)

José R. L. Neto Sirotto[a], Victor H. T. Soares[b] , Julio A. Cordioli, Rudner L. Queiroz[c], César J. Deschamps, Rafael Cooke, Leopoldo P. Bastos

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Abstract

Jet noise is one of the main noise sources in an aircraft, especially at take-off conditions. Considerable resources have been applied to the study of the noise sources mechanisms involved and ways to reduce it. Nevertheless, many questions still remain about jet noise generation and applicable noise mitigation techniques. In order to reduce the costs involved in jet noise studies, the use of experimental data obtained by means of measurements in smallscale models has been the norm. The so-called jet noise rigs aim at reproducing the noise generation phenomena present in real aircraft engine jets and, in order to be used with confidence, need to be properly design and its data validated. This work focuses on the acoustic validation of a jet noise rig recently built at the Laboratory of Acoustic and Vibration (LVA), Federal University of Santa Catarina (UFSC). To this aim, results from noise measurements in the new rig are compared with experimental data available in the literature and analytical models. Prior to the jet noise measurements, a detailed analysis was performed to

investigate relevant parameters that could affect the measurements, such as the acquisition system, positioning of microphones and background noise due to machinery. The validation considered the jet noise from a smooth circular nozzle with 2” diameters, cold flow and subsonic conditions (Mach number 0.3 to 0.9), and a chevron nozzle in the same conditions. In both case, the nozzle geometries used are the same as the series of NASA Small Metal Chevrons. Results indicated that LVA/UFSC jet noise rig can be considered acoustically validated, since the sound field measured is in agreement with published data and expected trends.

Tuesday morning, 6 September 2016 **Lounge Lateral Room**
10:00 - 10:40
POSTER SESSION - Monitor 2
Noise: Sources and Control
NS5 - Sustainable materials for sound absorption and insulation

POSTER

Sustainable Materials for Sound Absorption and Insulation:

Paper ICA2016-695

Acoustic properties of green walls: Absorption and insulation

Rodolfo Thomazelli[(a)], Fernando Caetano[(b)], Stelamaris Bertoli[(c)]
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Abstract

In the past few decades, the issue of environmental quality in the urban environment is a recurring theme. Green walls and roofs have demonstrated great efficiency in the attenuation of adverse effects such as heat islands. The thermal characterization of these systems are well studied, however few studies have been carried out on their acoustic potential. The characterization of the acoustic performance of green walls may provide better results for the acoustic simulation of urban environments where such elements are employed. This paper presents experimental results of the sound absorption and insulation tests for a modular system of vegetable panels. Each module was composed of a baseplate made of laminated plywood, equipped with 9 geotextile bags in which was inserted a highly porous substrate, and grown a specie of plant with high leaf density. Absorption tests were conducted in a reverberation room, where sound absorption coefficient over frequency was measured. The sound absorption of the panels were analysed in three different situations, in order to better characterize the contribution of each part of the system: (I) baseplate + geotextile bags, (II) baseplate + geotextile bags + substrate, and (III) plate base + geotextile bags + substrate + vegetation. Façade insulation tests were performed with the same types of sample placed on a concrete block façade of a small building. Results showed a significant increase of sound absorption coefficients at the whole spectrum when substrate and vegetation were inserted on the baseplates in each situation. Both the standardized level difference and the weighted standardized level difference showed a small improvement in the insulation for the sample with vegetation.

POSTER

Sustainable Materials for Sound Absorption and Insulation:

Paper ICA2016-865

Biobased porous acoustical absorber made from polyurethane and waste tires particles

Guillermo Soto[(a)], Nilda Vechiatti[(b)], Norma Marcovich[(a)], Federico Iasi[(b)], Mirna Mosiewicki[(a)], Alejandro Armas[(b)]

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Abstract

The production of flexible polyurethane foams (FPF) incorporating bio/recycled raw materials is an interesting alternative to conventional acoustic absorbent materials. In this sense, bio-based polyols like glycerol or hydroxylated methyl esters derived from tung oil as multifunctional polyols, and waste tires particles as fillers with capability for acoustical absorption and low thermal conductivity, are prospective feedstocks for FPF preparation. In this work, FPF were prepared by adding different amounts of these components to a formulation based on a commercial polyether polyol. Results of normal sound absorption coefficient measurements at different frequencies, scanning electron microscopy analysis and compression tests are presented and discussed. The addition of waste tires particles or glycerol to the commercial formulation gives good performance as acoustic absorbers from 400 500 Hz, with NRC and SAA values near and above 50%. Moreover, the absorption coefficient reaches high values mostly at the highest evaluated frequencies (~62-89% at 2000 Hz and ~70-91% at 5000 Hz), for 30 mm thickness samples. On the other hand, the obtained FPF presented enhanced both the modulus and yield stress and in all the cases, a high recovery of the strain (>90%) applied in compression tests was attained after 24 hours. Scanning electron microscopy micrographs revealed that the obtained foams present a combination of open and closed cell structures and both, the modifiers and particles, tend to decrease the cell size. Based on acoustical and mechanical performance, and morphological analysis, the results show that these new kind of materials could be innovative sound absorbers, cheaper and environmentally more convenient in comparison with other available materials.

Tuesday morning, 6 September 2016 **Lounge Lateral Room**
11:00 - 12:00
POSTER SESSION - Monitor 2
Noise: Sources and Control
NS4 - Materials for Noise Control

POSTER

Materials for Noise Control:

Paper ICA2016-128

Investigation of the noise reduction provided by vegetation belt in different design styles

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Abstract

Contemporary principles on Landscape plants design mainly guided by Visual effects, it's rarely reported that how the community structure influenced on noise attenuation. In this article, what effect did the community structure with Visual oriented design on noise attenuation was studied by researching the relationship between the arrangement of the green belt and noise reduction(LAeq and 1/3 OCT analyses) in Tianjin, China. 12 green belts were selected for the measurement of noise attenuation in Tianjin Avenue. Based on actual traffic noise as a source of noise, both LAeq and 1/3OCT were measured on both sides of the green belt, in winter and summer respectively. The results showed that the greatest effect of green belt structure on LAeq attenuation was the width of green space, especially the width of the woody pants layer. A parabola pattern was showed between the relationship of noise

attenuation and frequency by 1/3 OCT analyses in all green belts measured. The frequency band of maximum attenuation was 500-1600Hz, with the frequency of peak attenuation at 630Hz.

POSTER

Materials for Noise Control:

Paper ICA2016-326

Experimental study on sound performance of microperforated panel with membrane-type acoustic metamaterials composite structure

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Abstract

Microperforated panel (MPP) absorbers have been widely used in noise reduction and are regarded as a promising alternative to the traditional porous materials. However, the absorption bandwidth of a single-layer MPP is insufficient to compete with the porous materials. In order to improve the sound absorption ability of the single-layer MPP, MPP with membrane-type acoustic metamaterials composite structure is introduced. Sound absorption properties of the MPP with membrane-type acoustic metamaterials composite structure are studied by the impedance tube experiment. Results show that Membrane cell can change the sound absorption of MPP by introducing additional absorption peaks and valleys. Membrane cell and mass blocks mainly increase the normalized surface resistance of MPP with membrane-type acoustic metamaterials composite structure. Excellent performance of sound absorbing structure will be obtained by rational design of microperforated plate, the size of film cell, the number, size and position of mass blocks.

POSTER

Materials for Noise Control:

Paper ICA2016-475

Evaluation and comparison of acoustic insulation of traditional and alternative masonry in Tucumán, Argentina

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Abstract

This paper aims to qualify and quantify the properties, in general, and acoustic insulation, in particular, of new alternatives masonry in Tucumán, Argentina, and its comparison with traditional ones. The goals are to investigate, identify and catalog alternative masonry developed as those traditional that can be found in the local environment. Evaluate those selected according to: their manufacturing process, energy cost, economic cost and physical, technological and acoustic insulation properties; compare them; develop conclusions and recommendations for use in building systems. The methodological strategy implemented was the "participatory Action Research. The results achieved are: identification of numerous alternative masonry compounds using different raw materials from urban solid waste and wastes from different industries; determining acoustic insulation properties of alternative masonry; comparison with traditional ones; determining whether or not to use. It was concluded that many alternative masonry have equal, or in some cases better than traditional, acoustic properties and resources used for development are environmentally friendly; in some cases are economically profitable too. It was determined that is necessary to generate awareness in the population as well in the industry sector to ensure that such alternative masonry compete in the in the construction market with traditional ones to achieve better building at a time of a sustainable environment.

Tuesday morning, 6 September 2016 **Lounge Lateral Room**
09:40 -10:00
POSTER SESSION - Monitor 3
Communication Acoustics
CA2 - Communication Acoustics

POSTER
Communication Acoustics:
Paper ICA2016-833

Evaluation of the frequency response of consumer headphones and its influence towards a binaural reproduction with HRTF individualization

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Abstract
Spatial audio technologies based on binaural techniques have sparked a renewed interest in recent years thanks to the spread of headphones listening using smart mobile devices. However, the quality of the headphones provided with smartphones or music players can be a determinant factor in the resulting binaural spatial sound sensation. In the first part of this paper, a perceptual test has been carried out to investigate the effects of the headphones frequency response over the azimuth localization in the horizontal plane. A virtual simulation headphone technique was employed to test seven different high and low quality headphones. Results indicate that much front-back confusion is produced in all cases and that some specific frequency bands have an important role on it and also a poor response in some high frequency bands can affect the lateral positions localization. In the second part of the paper a technique based on modelling the frequency response of the headphones and the HRTF using a chain of peak filters is presented. This technique is suitable to be employed for the individualization of the HRTF and it will solve some of the localization problems commented using less equipment than other published individualization methods.

Tuesday morning, 6 September 2016 **Lounge Lateral Room**
10:00 - 10:40
POSTER SESSION - Monitor 3
Signal Processing in Acoustics
SP1 - Acoustic Array Systems

POSTER
Acoustic Array Systems:
Paper ICA2016-295

Development of a 3D impulse response interpretation algorithm

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Abstract
In this paper, the development of software 3D interpretation impulse responses is presented. This new tool allows to identify the spatial provenance for each time window of an impulse response recorded in a soundfield microphone. In this way it is possible to discern between the direct sound and the different reflections both temporarily and spatially. To understand the basics of this method, a brief introduction to all concepts involved is presented. Alternatively, different types of procedures are shown for data evaluation. Finally system applications are shown in controlled acoustical environments so evaluating the precision and the potential of this tool.

POSTER

Acoustic Array Systems:

Paper ICA2016-352

A sound source localization algorithm using microphone array with rigid body

Tao Song[(a)], Jing Chen[(a)], DaiBing Zhang[(b)], Tianshu Qu[(a)], Xihong Wu[(a)]

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[(b)] Colllege of Mechatronic Engineering and Automation, National University of Defense Technology, Changsha, China

Abstract

In general, most microphone arrays use the time differences between the different microphones to localize the sound source. The level differences between the different microphones always are neglected. For using both of them, this paper propose a sound source localization algorithm based on the direction transform functions from the spatial points to each of the microphones which are attached on the rigid body. Firstly, the signals recorded by each microphone are filtered by the inversed direction transfer function; Secondly, the cross-channel similarities between each pair channel signals are calculated to construct the similarity matrix; Lastly, the mean variances of the similarity matrix of all the test directions are calculated and the sound source is localized in the direction which the minimal variance correspond to. The proposed algorithm is evaluated theoretically at signal-to-noise ratios (SNR) from 40 to -40dB. In the evaluation experiments, there are six microphones attached on the surface of a rigid ball evenly, and their direction transfer functions are calculated using the spherical head model. Three types of signal, Gaussian white noise, speech and music, are used as test stimulus. The evaluation experiment's results show that the steered response power of the proposed method performs better than that of the phase transform algorithm (SPR-PHAT).

Tuesday morning, 6 September 2016 **Lounge Lateral Room**
11:00 - 11:40
POSTER SESSION - Monitor 3
Signal Processing in Acoustics
SP3 - Model-Based Optimization/Estimation and Analysis

POSTER

Model-Based Optimization/Estimation and Analysis:

Paper ICA2016-367

An artificial neural network model for predicting sound direction in different acoustic environments

Tao Song[(a)], Tianshu Qu[(a)], Xihong Wu[(a)], Jing Chen[(a)]

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Abstract

Interaural time difference (ITD) and interaural level difference (IID) are two major cues for binaural sound localization. Traditionally computational models predict sound direction by extracting ITDs and IIDs directly, which could not work well when the acoustic environment was changed, e.g., from an anechoic room to a reverberant room. In this work, a new model based on artificial neural network was built to adapt to different acoustic environments. The input is binaural signals. For each ear, the signal is firstly filtered into 18 frequency bands according to 1/3 octave bands. In each band, the signals from both ears are input into a three-layer neural network. Units representing neurons are distributed on a delay line for each ear. At the first layer, the ITD and IID of each unit are calculated and represented. The inhibition processing is simulated at the second layer to enhance the output of the first layer. At the third layer, the units are mapped to different spacial locations, and the output is the firing probability of each unit. Outputs across frequency bands are combined by a weighted summation, where the weights are determined at the training stage. Finally, the unit with the maximum firing probability indicates the sound direction. Two experiments were conducted to testify the validity of the model in anechoic and in reverberant environment, respectively. Stimuli used as training data and in tests are all generated by

filtering Gaussian white noises with head related impulse responses. The experimental results show the percent correct of localization is 100% for the anechoic condition, and 98.9% for the reverberant condition. The error is analyzed and the limitation of this model is discussed.

POSTER

Model-Based Optimization/Estimation and Analysis:

Paper ICA2016-530

A new toolbox for the identification of diagonal Volterra kernels allowing the emulation of nonlinear audio devices

Thomas Schmitz[a], Jean-Jacques Embrechts[b]
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Abstract
Numerous audio systems are nonlinear. It is thus of great importance to study them and understand how they work. Volterra series model and its subclass (cascade Hammerstein-Wiener model) are usual ways to modelize nonlinear systems. However the identification methods of these models are still considered as an open topic. Therefore we have developed a new optimized identification tool ready for use and presented as a Matlab toolbox. This toolbox provides the parameters of the optimized sine sweep needed for the identification method, it is able to calculate the parameters of the Hammerstein model and to emulate the output signal of a nonlinear device for a given input signal. To evaluate the toolbox, we modelize a guitar distortion effect (the Tubescreamer) having a total harmonic distortion (THD) comprised in the range 10-23%. We report a mean error of less than 0.7% between the emulated signal and the signal coming from the distortion effect.

Tuesday midday, 6 September 2016 **Juan Pablo II Auditorium**
12:00 - 13:00
Plenary lecture:
Chair: Michael Taroudakis

Chen-Fen Huang

Paper ICA2016-485

On the perspective of underwater acoustic tomography for probing ocean currents in shallow-water environments

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Abstract
Oceanographic processes in coastal regions including wind driven flows, tidal currents, river outflows, internal waves, eddies, western boundary currents, etc. are highly variable in time and space. Conventional oceanographic measurements (e.g., acoustic Doppler current profiler) cannot provide a synoptic image of those dynamic processes, especially for short time and space scales. Ocean Acoustic Tomography (OAT) uses time-of-flight measurements from different angles across the water. OAT is an effective method for mapping the spatial distribution of current and temperature fields. This talk will focus on the OAT applications to probe the current field in shallow water environments and present recent experimental results. Included are 1) the application of the middle-range (~50 km) OAT technique to study the spatial and temporal variations of the sub-branch of the Kuroshio off the east coast of Taiwan, 2) exploiting the communication signals of distributed networked underwater sensors for ocean current mapping, and 3) integrating moving vehicles to enhance OAT results.

Noise Assessment and Control:

Paper ICA2016-105

Alternative approach for valuing road traffic noise: Case study of Quito, Ecuador

Luis Bravo-Moncayo[(a)], José Lucio-Naranjo[(b)], Ignacio Pavón[(c)]
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Abstract

This study reports the results of a contingent valuation of road traffic noise in Quito, Ecuador. Two approaches were evaluated: an ordered probit econometric model and a committee of artificial neural networks. The input variables were obtained through a social survey that assess the respondent's noise perception and socio economic status as well as the modelled noise exposure level. A comparison indicates that the ANN model can predict the willingness to pay to reduce road noise annoyance with 85.7% better accuracy than the econometric model. The proposed approach reaches an adequate generalisation level, becoming a tool for determining the cost of transportation noise in a policy-making context.
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Noise Assessment and Control:

Paper ICA2016-247

Influence of the temporal and spectral structure of the road traffic noise on annoyance

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Abstract

The relationship between road noise annoyance and the acoustic descriptors A-weighted equivalent sound pressure level [LAeq], temporal sound level variance [TSVL], crest factor [CF] and spectral centroid [G] was studied. Road traffic noise was measured and recorded. The acoustic descriptors were obtained from these noise samples. Noise annoyance was evaluated by a socio-acoustic survey performed under laboratory conditions on a representative population sample of Quito. The questionnaire assessed noise annoyance, noise sensitivity, demographic and socioeconomic data. Through Pearson's correlation coefficient it was shown that noise sensitivity predicts noise annoyance. ANOVAs found that the acoustic descriptors considered were an influence on noise annoyance. The descriptors that predicted annoyance most accurately were LAeq and G. Fluid continuous traffic flow was found to be the most annoying, while intermittent traffic flow was the least annoying. It was found that highly sensitive people tend to use private cars, prefer intellectual jobs, lack concentration, and experience headaches, sleep disturbance and irritability. They are also more likely to have high levels of daily stress. The possible relationship between sensitivity and the psychological trait of negative affect is discussed. A multinomial ordered probit regression was used for a deeper analysis of annoyance. The four descriptors, as well as traffic type, age, the presence of noise at childhood and noise sensitivity were all influential on the reported noise annoyance. A binary extreme value model was performed for noise sensitivity; gender, lack of concentration, sleep disturbance, irritability, type of work, preferred means of transportation and daily reported stress led to changes in noise sensitivity.

Noise Assessment and Control:

Paper ICA2016-390

Noise observatories in French cities. 20 years of practice. Objectives, issues and methods

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Abstract

For the last 20 years, the non-profit organisation Acoucité located in the Greater Lyon (South- Eastern France) has been a noise skill centre serving urban areas. It has led the development of sound environment observatories in France and Europe and conducted research programmes in partnership with its founding members (public research centres and local authorities). This article aims to provide a feedback on the characteristics of noise observatories (missions, objectives, issues, resources, levers, and changes) and to present the major research programmes in which Acoucité has participated. The observatories must respond to four needs: describing the existing situation, tracking changes, anticipating changes, and finally spreading knowledge and the development of innovative solutions. They must also stand out from a metrological-centred approach to integrate sensory criteria such as perception (e.g. discomfort) and public expectations (citizens, policymakers, elected officials, etc.). A sound environment observatory helps to develop tools and methods oriented towards tracking and monitoring urban changes through the levers of public policy. This article offers a descriptive approach to these practices. Recent technological developments such as open data should prompt questions and open up new ways to both gather and circulate transparent, relevant and easy to understand noise information. Finally, as interests for economies of scale and complementarities arise, it is also necessary to consider a relationship with other environmental fields such as air quality. This article offers to highlight these polymorphous and multidisciplinary issues, through a scientific and technical approach that is nevertheless pragmatic, and even sometimes empirical.

Noise Assessment and Control:

Paper ICA2016-429

Applying low noise road surfacing to reduce road traffic noise in Hong Kong

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Abstract

Hong Kong is a very small but hyper-dense city with over 7 million people living in 1,100 sq. km in which 85% of land is hilly area. Due to the need to build residential developments to accommodate the population and at the same time, the need to have very concentrated road network systems to support economic growth, residential buildings are unavoidably built next to highways and road traffic noise in Hong Kong is undoubtedly the major environmental noise problem affecting residents. The Government of Hong Kong is committed to tackle road traffic noise problem and adopts paving of low noise road surfacing (LNRS) as one of practical solutions to reduce the traffic noise impact, especially in the busy urban districts. Researches and trials of different forms of LNRS materials, such as friction course, polymer modified friction course, crumble rubberized asphalt, polymer modified stone mastic asphalt under different traffic conditions have been conducted to evaluate the engineering durability and noise reduction ability. This paper reviews and presents the onsite experiences and the technical findings including noise reduction effectiveness, engineering durability etc. of these materials in Hong Kong traffic situations.

Noise Assessment and Control:

Paper ICA2016-901

Special acoustic area: The Center of the Malaga City (Spain)

Fernándo López[a], David Carretero[a], Isabel Gimenez[a], Ricardo Hernández-Molina[b]
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Abstract

The definition as special acoustic area in the center of the city Malaga is determined by a study that looked at the results of the strategic noise map of the city, treating more than 20,000 complaints about noise pollution, noise monitoring established in the city and an exhaustive study of the different acoustic sources existing in the area. Diagnosed noise pollution need a specific zonal plan to achieve the objectives of acoustic quality. The present paper develops methodology of analysis applied with particular emphasis on quantifying noise sources leisure generated by the concentration of people in the street.

Tuesday afternoon, 6 September 2016 **Cardenal Pironio Auditorium**
16:10 - 16:40
Environmental Protection Agency
Ministry of Environment and Public Space.
Strategic noise map of the city of Buenos Aires

Tuesday afternoon, 6 September 2016 **Juan Pablo II Auditorium**
16:30 - 17:50
Environmental Acoustics & Community Noise
EN1 - Noise Assessment and Control

Noise Assessment and Control:

Paper ICA2016-79

Noise action planning in Germany

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Abstract

In 2002, to improve the noise situation in Europe, the EU issued the Environmental Noise Directive (2002/49/EC). It became law in Germany in 2005. The aim of the Directive is to reduce environmental noise and to prevent an increase in noise in areas which are traditionally quiet. This first requires mapping the level of noise pollution in different areas and then introducing specific measures to reduce it. The EU Directive envisages progressive implementation of its provisions. Since 2012, noise levels in all conurbations and on all major transport routes in Europe need to be recorded. In Germany, this involves 71 conurbations with around 24.5 million inhabitants, 44,000 kilometres of motor-ways and major trunk roads, 13,700 kilometres of major railway lines, and all eleven major airports. Moreover, effective noise abatement measures had been established in Noise Action Plans. The state of all Noise Action Plans until 2015 were analysed. Experiences, promising approaches and difficulties were compiled. A report provided guidance for the improvement of Noise Action Plans and their general conditions.

Noise Assessment and Control:

Paper ICA2016-629

Data analysis of noise complaints in Región Metropolitana, Chile

Ismael Gómez[a] [b], Max Glisser[a] [c], Camilo Padilla[a]

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Abstract

According to the First Environmental Perception Survey performed by the Chilean environmental authority in 2015, noise is the second greatest environmental problem in Santiago de Chile. Even though the road traffic noise, one of the main urban noise sources, has been quantified by the *Ministerio del Medio Ambiente* (MMA) with the *Gran Santiago* noise map, and the noise sources regulated by the *Decreto Supremo* N°38/2011 of MMA are controlled by the *Superintendencia del Medio Ambiente*; the "behavioral" noise sources, according to the *Sistema Nacional de Información Ambiental* definition, are regulated by Local Policies of each Municipality and formalized first mainly through *Carabineros de Chile* (CCh), and often this data is not included in official documentation. A more complete analysis of the noise complaints received by the institutions involved in his management allows identifying critical sectors, possible causes and solutions. For this reason, this study contains an analysis of the noise complaints received by CCh in *Región Metropolitana* between 2010 and 2015, submitting the total number of complaints, the monthly mean and the distribution by municipality, both in a 6 years period as well as for each year individually. The link to source code used for the process and analysis of data developed with the free access software "R" and its packages is presented. The information used in the analysis was requested by *Ley de Transparencia*.

Noise Assessment and Control:

Paper ICA2016-417

Comparison of noise pollution complaints concentration mapped in three capitals of Brazilian northeast

Luciana Alves[a], Tamáris Brasileiro[b], Renata Araujo[c], Débora Florêncio[d], Lorena Firmino[e], Caio Almeida[f], Bruna Alencar[g], Maria Lúcia Oiticica[h], Virgínia Araújo[i], Bianca Araújo[j]

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Abstract

The big cities of Brazil, especially the major ones, have similar characteristics in relation to environmental problems. Environmental noise is a physical phenomenon that directly affects the population and is classified as noise pollution when presenting sound levels above the desirable to health and acoustic comfort. Confirmation of the existence of noise pollution in a city is usually accompanied by coercion actions to stop the problem, whereas it should trigger preventive actions to reverse the situation, considering that the population itself is both causer and victim of the problem. Its only defense is the complaint to government agencies. Therefore, it is necessary to understand the triggering event and the location of the sound sources to then think about noise pollution prevention actions in cities. In this scenario, this research seeks to characterize noise in João Pessoa/PB, Maceió/AL and Natal/RN cities, all in Brazil's northeast, in the last four years (2012-2015), through the comparative analysis of concentration mapping of noise pollution complaints. From the results, it is concluded that a) from environmental pollution, noise is the highest percentage of complaints in João Pessoa and Natal, in contrast to Maceió, which in recent years, fell to second place; b) João Pessoa has the highest rates of complaints of noise, followed by Natal and Maceió; c) in relation to the most common sound sources, stand out in the three cities, speakers and live music are more recurrent,

although it was found that in Natal and Maceió such sources are from bars, while in João Pessoa those originate from nuisance between neighboring; d) despite the culture of complaints are distinct, the three cities needs environmental education to mitigate the noise pollution levels.

Noise Assessment and Control:

[Paper ICA2016-657]

Noise pollution in Fortaleza, synonymous with discomfort, sex and violence in urban areas

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Abstract

Fortaleza is a Brazilian city of about 3,000,000 people, 900,000 vehicles, airport in the central area, subway with surface section, recreational facilities with sound events, but mostly it was the place of origin of cars with powerful sound attached vehicles, known as sound walls. grotesque idea that came to end with the tranquility of the city and be pivotal to violence and youth sexual behavior degeneration niches, adding to the already noisy urban profile of the city, noise levels exaggerated and dangerous social aspects. This work shows the new face of the city with the introduction of these intruders and details what is being done to combat the problem and the results obtained

Tuesday afternoon, 6 September 2016 **Dr. Valsecchi Auditorium**

14:30 - 16:10

Architectural Acoustics - Room and Building Acoustics

AA5 - Challenges and Solutions in Acoustical Measurements and Design

Challenges and Solutions in Acoustical Measurements and Design:

[Paper ICA2016-169]

Effect of sample area in reverberant chamber measurements of sound absorption coefficients

António P. O. Carvalho[a], Mário R. M. Sousa[b]

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Abstract

To evaluate the influence of the sample size on the experiments in a reverberant chamber to determine the sound absorption coefficient, the reasonable minimum sample area needed was analyzed, considering that the quantity of material is important for those who want to test it. Samples from 1 to 12 m^2 (by increments of 1 m^2) were used. The influence of the sample position on the room's floor has also been tested for a 10 m^2 sample arranged on the center of the reverberating chamber and scattered around the room.

Challenges and Solutions in Acoustical Measurements and Design:

[Paper ICA2016-238]

Characterization of sheep wool panels for room acoustic applications

Umberto Berardi[a], Gino Iannace[b], Maria Di Gabriele[c]

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Abstract

Given their good thermal insulation properties, sound absorption behaviour, lack of harmful effects on health, and availability in large quantities, natural fibers are becoming a valid option for sound absorption panels in building applications. This paper presents the characterization of sheep wool fibers and panels. The absorption coefficient and the static flow resistivity for samples of different thickness have been measured. It has discussed the possibility of using fabrics obtained with different kinds of woven wool as sound absorbing systems. For this scope, wool tapestries were mounted at a variable distance from the rigid back wall. The high absorption obtained in some frequency bands, depending on the back cavity depth, confirmed the possibility of use wool tapestries for ad-hoc customized acoustic interventions. Finally, this paper discusses the advantages to adopt sheep wool for room acoustic applications.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-205

Measurement and a new model of resonances of Helmholtz resonators and tubes

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Abstract
It is known that a Helmholtz resonator can be written as series resonance circuit with a capacitor and a coil. But by measuring frequency characteristics of the entrances of a Helmholtz resonator and tubes, it is shown that their frequency characteristics have a zero and a pole frequencies, in a pair. Next, we have measured inside and outside, sound pressures of the Helmholtz resonator and tubes, by probe microphones and microphones in detail. We calculate their amplitudes and phases from correlation coefficients by FFT. It is shown that their zero frequencies are resonance ones of series resonance circuit with a capacitor and a coil, and that their pole frequencies are lower than their zero frequencies. It is shown that their pole frequencies are almost the same as their pole frequencies when their outside bottoms are tapped. We propose equivalent circuit models of Helmholtz resonators and tubes.

Challenges and Solutions in Acoustics Measurement and Design:
Paper ICA2016-809

Façade insulation at low frequencies – influence of window design

Krister Larsson[a], Dag Glebe[b]
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Abstract
Exposure to environmental noise in the neighbourhood has negative effects on the wellbeing and the quality of life of residents. Protection from environmental noise and keeping indoor noise at acceptable levels are therefore essential properties of building facades. The sound insulation of a façade depends not only on the design of the wall elements, but on the combination of all components and their assembly, such as windows, air terminals, seals etcetera. However, windows are often the weakest component and determine the sound insulation. Energy demands, as well as building cost and sustainability demands, lead to the development of new building elements and constructions, often using lightweight solutions. In many cases the low frequency (<200 Hz) sound insulation is a challenge for lightweight constructions, and the resulting indoor levels also depend on the source spectrum. Additionally, the low frequency sound insulation is not only a characteristic of the separating element itself, but depends on room design and modal behaviour. Although A-weighted indoor levels meet requirements, residents may be annoyed by low frequency noise and it might be difficult to meet additional low frequency demands. In this paper the challenge of low frequency sound insulation in modern single- and multi- family houses in Sweden are studied theoretically and experimentally. The influence of window design on the resulting indoor levels is presented for various environmental noise sources for typical room sizes.

Challenges and Solutions in Acoustics Measurement and Design:
Paper ICA2016-822

Façade insulation at low frequencies – influence of room acoustic properties

Dag Glebe[a], Krister Larsson[b]
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Abstract
Exposure to environmental noise in the neighbourhood has negative effects on the wellbeing and the quality of life of residents. Protection from environmental noise and keeping indoor noise at acceptable levels are therefore essential properties of building façades. The sound insulation of a façade depends not only on the design of the wall elements, but on the combination of all components and their assembly, such as windows, air terminals, seals etc. However, windows are often the weakest component and determine the sound insulation. Energy demands, as well as building cost and sustainability demands, lead to the development of new building elements and constructions, often using lightweight solutions. In many cases the low frequency (<200 Hz) sound insulation is a challenge for lightweight constructions, and the resulting indoor levels also depend on the source spectrum. Additionally, the low frequency sound insulation is not only a characteristic of the separating element itself, but depends on room design and modal behaviour. Although A-weighted indoor levels meet requirements, residents may be annoyed by low frequency noise and it might be difficult to meet additional low frequency demands. The influence of window designs on façade insulation is evaluated in a corresponding paper, and the scope is here broadened to include room designs. In this paper, the low frequency interaction between the source spectrum, the façade insulation and various room acoustic properties is evaluated numerically and experimentally. In particular, the influence of various absorber configurations on the modal behaviour is discussed, in relation to corresponding measurement challenges in the low frequency region.

Tuesday afternoon, 6 September 2016 **Dr. Valsecchi Auditorium**
16:30 - 18:50
Architectural Acoustics - Room and Building Acoustics
AA5 - Challenges and Solutions in Acoustical Measurements and Design

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-552

Noise measurement and mitigation for urban building foundation excavation

Jack B. Evans
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Abstract
Excavation noise disturbance case studies are presented for two urban locations; i) a large academic building near a hotel-conference centre and ii) a high rise office building adjacent to an owner-occupied residential condominium, respectively. Impact, milling and drilling in hard limestone resulted in noise and vibration disturbance complaints from nearby buildings. In addition to airborne sound intrusion via windows and walls, there were indications of reinforcing re-radiated interior sound from ground borne vibration transmission into building structures. The acoustical design objectives were to a) determine noise levels of excavation machinery, impact and scraping, and b) to mitigate noise intrusion into nearby buildings to meet criteria limits and to reduce occupants' distraction and annoyance. Permissible property boundary noise limits exist, but noise limits at building façades relating to interior background sound levels were controlling. Noise reductions were essentially limited to existing insulating glass window performance. Excavation noise contains machinery engine noise, repetitive ram-hoe impacts on rock, scraping noises of milling machine and line drills. The impact procedures transmit potentially "feelable" vibration. Dropping excavated soils and debris into trucks for hauling causes audible impacts. Short-term on-site noise monitoring determined noise spectrum levels and variation over time. Measurement results are summarized in graphs, including statistical (Ln) percentile sound pressure levels (dB) in 1/3-octave spectra. Limited interior floor vibration from one building will be included for comparison. Photos will illustrate excavation equipment and other relevant conditions. Noise mitigation recommendations are summarized.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-484

The effects of the excitation source directivity on some room acoustic descriptors obtained from impulse response measurements

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Abstract
Reverberation time, early decay time, center time, clarity, and definition are classical objective metrics to assess the acoustic performance of closed spaces. These parameters can be evaluated from the room impulse response, which is usually estimated through measurements of the sound pressure field produced by an omnidirectional loudspeaker array inside the room. However, typical sound sources in a real situation will never be omnidirectional. Therefore, the question arises whether or not the so-obtained room acoustic descriptors are still meaningful for different source directivities. In order to shed light on this question, we conducted a set of measurements using a directivity controlled sound source, which is a compact array of independently driven loudspeakers, so that it can operate as a monopole, dipole or quadrupole source by controlling the signals sent to the loudspeakers. The impulse responses of two class-rooms of the same volume – with and without acoustic conditioning – and the corresponding room descriptors were experimentally obtained for different source directivities, namely, mono-pole, dipole with several distinct orientations and a quadrupole. We observed that the directivity affects significantly the room acoustic parameters, regardless of the acoustic conditioning. Also, the descriptors more sensitive to directivity changes were those related to the balance between early and late arriving energy, such as clarity, definition, and center time. These results show that the directional characteristic of the sound source plays an important role in room acoustics, and thus it cannot be neglected when designing and/or assessing a closed space.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-21

Measuring sound insulation from partitions through a deconvolution technique

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Abstract
The most popular and traditional method for measuring sound insulation of partitions, as described in ISO 10140, is performed through a sound level meter with a broad band random excitation signal. Additionally, the reverberation time in the receiving room must be measured, frequently through the interrupted noise method. However, alternative ways for evaluating the sound pressure level in source and receiving rooms, and for measuring the reverberation time, are available for decades. Some of these alternatives are based on the measurement of transfer functions through correlation or deconvolution techniques, which supposedly lead to more precise results. This paper presents the foundations for using a deconvolution technique in such a context, and comparisons from experiments performed with the traditional and the deconvolution technique. Such experiments were designed to introduce different background noise conditions and the presence of air flow (which should violate the system's invariant condition for using the deconvolution technique). Results show that the uncertainties associated with the alternative method tend to be slightly lower than those observed from measurements performed with the traditional method. Comparison from a repeatability investigation shows that the alternative method is significantly more precise than the traditional one in regard to this aspect.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-831

A Study on acoustic impedance in-situ measurement technique using cardioid microphones

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Abstract

For acoustic impedance or absorption coefficient measurement of materials, P-P (two omnidirectional microphones) or P-U sensor are often used. Both methods have some problems. If P-P sensor is chosen, it has to be closed to boundary of a material. Therefore, the resolution of low frequency is less than that of high frequency. If P-U sensor is chosen, it requires great care and the calibration at short intervals. The authors have developed sound pressure and velocity esti- mating method using two cardioid microphones. In this report, the possibilities of using cardioid microphones for acoustic impedance measurement are discussed through the results of using P- P sensor and the proposal method. The results shows us that two omnidirectional microphones can measure up to 4 kHz. On the other hand, two microphones can measure over 4 kHz up to 10 kHz or more over. These results tell us that two cardioid microphones have advantage to two omnidirectional microphones.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-224

Results of measurements room acoustics properties of open plan offices based on parameter A-weighted sound pressure level of speech

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Abstract

In open plan office rooms, employees should be able to verbal communication with other employees located at the neighboring work stations, but their conversations should not disturb other employees. In order to assessment acoustic properties of the open plan office rooms should be used parameters from standard EN ISO 3382-3. These parameters based on two main parameters obtained from measurement: A- weighted sound pressure level of speech and speech transmission index STI. The first one main parameter is used to calculate parameters: spatial sound distribution of the A- weighted sound pressure level of speech, spatial decay rate of speech $D_{2,S}$ and A- weighted sound pressure level of speech at a distance of 4m (from the speaker) $L_{p,A,S,4m}$. The second one main parameter is used to calculate parameters: spatial speech transmission index STI, distraction distance r_D and privacy distance r_P. This paper presents results of tests and acoustic assessment parameters calculated from the first main parameter. Is used measurement methods and criterial values from the standard EN ISO 3382-3. Result of calculation: A-weighted sound pressure level of speech at the distance of 4 m was 43.8–54.7 dB (maximum permissible value - 48 dB; i.e. two out of six rooms fulfill satisfy the criterion), spatial decay rate of speech $D_{2,S}$ was 1.8–6.3 dB (minimum permissible value - 7 dB; i.e. none of the rooms fulfill satisfy the criterion). Result of measurement confirms previous studies of the author, that the acoustic properties of open plan offices rooms do not fulfill the criteria specified in the standard EN ISO 3382-3.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-537

How to reduce uncertainties for building acoustic measurements in test facilities?

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Abstract
Measurements in building acoustics are performed for very different purposes. One main purpose is to determine the acoustic properties of building elements in test facilities. For these measurements, a small uncertainty is desired by manufacturers and laboratories to discriminate between products. To achieve small uncertainties, test conditions are more and more specified, e.g. by defining the test geometry with glazing and windows. This approach has the disadvantage that the sound insulation in different geometries will distinguish from the one under specified conditions. Another approach to reduce the uncertainty is to define reference objects for specific types of building elements. The acoustic property of this reference object is then measured in a laboratory. When the measurement result is within specified tolerances, the laboratory is permitted to report an uncertainty smaller than the general uncertainty stated in ISO 12999-1. Both approaches and their implications are discussed in the contribution.

Challenges and Solutions in Acoustical Measurements and Design:
Paper ICA2016-451

Experimental measurements of flanking transmission in CLT structures

Luca Barbaresi[(a)], Federica Morandi[(b)], Massimo Garai[(c)], Alice Speranza[(d)]
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Abstract
This paper discusses the results of an experimental campaign that lead to the characterisation of flanking transmission in CLT structures with different connection systems. The vibration reduction indices Ki j were measured in accordance with the ISO 10848-1 standard in order to provide data suitable to calculate the apparent reduction index according to the EN 12354-1 standard. The research explored several vertical and horizontal junctions on which different number and the type of screws and plates, different CLT panels, and resilient materials at the wall-floor junction were tested. The analysis of these data will hopefully provide a significant tool to improve the accuracy of the prediction methods and to help the acoustic designer of timber building.

Tuesday afternoon, 6 September 2016 **Cardenal Pironio Auditorium**
15:30 - 16:10
Structural Acoustics and Vibration
SV1 - Structural Health Monitoring and Sensor Networks

INVITED
Structural Health Monitoring and Sensor Networks:
Paper ICA2016-654

Applications of acoustic emission monitoring to the assessment of structural integrity of rocks

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Abstract
At microscopic level, natural pre-existing flaws in rocks act as local stress concentrators, favoring the propagation of microcracks, resulting in a loss of structural integrity and brittle failure of rocks. Crack propagation in rocks causes a sudden release of energy as transient elastic waves known as acoustic

emission (AE). AE monitoring enables an insight into the cracking process without affecting the integrity of the specimen. In this work, marble and monzogranite specimens were subjected to displacement-controlled diametral compression tests. The descriptions of the loss of structural integrity and the failure process of these rock specimens on a scale ranging from microscopic to macroscopic were based on: three-dimensional localization of AE sources, AE parameters and signal processing in both the time domain and the frequency domain. Such descriptions were complemented with petrographic analyses, interpretation of the load versus displacement curves and visual examination of the specimens. Microcracking in monzogranite and marble was initiated at approximately 90% and 60% of the peak load, respectively. Before peak load, both rocks showed the development of microcracks uniformly distributed on the plane containing the loading edges. After the peak load, cracking extends through the thickness of the specimen all the way to the other face. The cracking process in monzogranite is progressive, brittle, explosive and localized, and involves a high release of AE energy. In contrast, marble exhibits a continuous, gradual and scattered cracking that releases low levels of AE energy in comparison with monzogranite. The analysis of AE signals in the frequency domain showed that the microcracking is characterized by wide band emissions.

INVITED

Structural Health Monitoring and Sensor Networks:

Paper ICA2016-857

Identification of dynamic response parameters of a concrete building during recent earthquakes by using structural vibration monitoring

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Abstract

The main building of Faculty of Engineering at Pontificia Universidad Católica de Valparaíso (Chile) has been instrumented with a vibration sensor network that consists of three synchronized tri-axial accelerometers installed on three different building's floors: base (-1), 2nd and 4th floor. Since September of 2015 to May 2016, this concrete building has been subjected to more than 400 sensible earthquakes with magnitude (MI or MW) between 4.0 and 8.4, and peak ground acceleration (PGA) between 0.006% and 5.41% of g. Previous to begin to operate the vibration sensor network, the soil and structure have been studied using microtremor data to identify the soil's resonant period and building's vibration periods. This work reports the main considerations for the configuration network; the pre-processing of each seismic vibration record; the analysis with technics of dynamic systems identification based on the time domain and on the frequency domain in order to obtain the vibration periods. With these results the time variation and the correlation of the vibration periods with seismic intensity index were studied. The vibration sensor network's configuration identified adequately the most sensible earthquakes but produces a lot of records of false earthquakes. In other hand, the techniques are not equally effective to their purpose, but all are applicable, in fact is recommendable to use two or more of this techniques simultaneously. No structural damages were observed due to the analysed earthquakes, but significant variations of the vibration periods of building were observed. Then, the time variation of periods is not necessary a good predictor to the damage. Finally, the use of the post-event period – pre-event period ratio was proposed as a better damage predictor.

Structural Acoustics and Vibration (others):
Paper ICA2016-44

Different input parameters in modelling for predicting impact noise of non-homogenous floors

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Abstract
Floor systems with non-homogeneous slabs have more complex means of propagation than homogeneous systems, with more variables to be considered in predictions by theoretical models. For those slabs, it is necessary to understand the differences of each material composing each subsystem, and the connection types between the elements of each one of this subsystem. Some floors integrating lightweight elements without structural purposes, are broadly used in several countries in precast slabs. The predictions based on computer modelling for building systems can be influenced by the input parameters related to connections between the elements of the floor system. In building structures, the analysis of radiation due to element vibrations may be represented by wave propagation relationships as a one-dimensional system, a two-dimensional system or a three-dimensional solid. In these floors the modelling of the interaction between elements can be basically a face, a line or a point connection. In addition, the choice of the connection type can determine the vibration transmission amongst all the floor elements. This study focuses on the differences that can be obtained in the induced vibration response due to an impact source on non-homogeneous slabs. It also presents some examples of modelling options for several floor systems, considering input parameters for different connection types.

Structural Acoustics and Vibration (others):
Paper ICA2016-78

Vibration isolation and seismic restraint for mechanical equipment in Chile

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Abstract
Vibration control of mechanical equipment in seismic countries is a subject that should be addressed with some caution. One of the most utilized elements in controlling vibration is a spring isolator, one that has high deflection (relative to neoprene isolators). The problem is that the resonant frequency can coincide with the disturbing frequency of an earthquake, which means that in an earthquake the spring will begin to ‘jump’, causing the displacement of the mechanical equipment. The displacement of the equipment, beyond generating high costs (damage, loss of use, flooding, etc.), also creates the risk of personal injury and even death. Due to the aforementioned risks, it is necessary to consider the seismic ‘variable’ when doing a proper vibration control design in a seismic country or region. Criteria, designs and recommendations are presented in this report in order to perform proper vibration isolation in conjunction with seismic restraint for mechanical equipment and systems in Chile. In addition, you will be shown some failures from the M8.8 earthquake on February 27, 2010 in Chile.

Structural Acoustics and Vibration (others):

[Paper ICA2016-80]

Study of the sound radiation of a rectangular plate resting on a winkler elastic foundation

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Abstract

In this work, the effect of a Winkler elastic foundation on the sound radiation of a rectangular plate is both analytically and numerically studied using the Resistance Radiation Matrix method. An exact solution of the damped natural frequencies of the free and forced vibration of the plate with different boundary conditions is presented. The effect of the stiffness of the elastic foundation on the sound power radiated by the plate is also investigated. It is concluded that the damped natural frequencies for both SS-C-SS-C and for a SS-SS-SS-SS plate, depend upon the stiffness of the elastic foundation. However, the mode shapes of the plate do not depend on this parameter. On the other hand, it is shown that, in the case of forced vibration, the sound power level of the plate has an inverse relationship with the stiffness of the Winkler elastic foundation. This feature is correctly described by the proposed mathematical model.

Structural Acoustics and Vibration (others):

[Paper ICA2016-10]

The effect of the cushion on reducing underwater noise caused by offshore pile driving

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Abstract

The underwater noise from offshore pile driving has been getting increasing attention due to its deleterious effects on marine animals. Two models are established to investigate the influence of the cushion parameters, including stiffness and restitution coefficient, on the underwater sound radiation. The first model, based on a nonlinear finite difference formulation, is used to compute the impact force on the pile head and the soil penetration for a single impact. The underwater sound pressure radiated from the cylindrical pipe pile can be predicted by the second model based on a semi-analytical variational formulation. For the examined pile, the cushion can dramatically reduce the high-frequency underwater noise. A proper cushion reduces the peak sound pressure level by 17 dB but hardly reduces the soil penetration depth for each impact. The research can provide noise reduction designing schemes for pile driving system by choosing appropriate cushions based on prediction of underwater noise and soil penetration.

Structural Acoustics and Vibration (others):

[Paper ICA2016-41]

Calculation of sound radiation in infinite domain using a meshless method

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Abstract

A meshless method coupling with a variable order infinite acoustic wave envelope element for sound radiation calculation in infinite domain is presented with the aim of accurately calculating the sound radiation and improving computational efficiency. It is based on using the element-free Galerkin method in the inner region enclosing the radiator and a variable order infinite acoustic wave envelope element in the outer region for the proper modeling of the pressure amplitude decay to satisfy the Somerfield condition explicitly. The details are provided for the derivation and implementation of this method. The factors of influencing the performance of the method, which include the shape function constructing, the weight functions, and the support domain, are discussed. The suitable radius of the influence domain for the acoustic field calculation in free space is also determined by use of numerical experiments. An infinitely long cylinder is designed for simulation to validate the method. The results illustrate the accuracy, applicability and effectiveness of this method.

Structural Acoustics and Vibration (others):

Paper ICA2016-176

Instability prediction of a fully updated brake system with uncertainty in contact conditions

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Abstract

Brake squeal, as friction-induced audible noise above 1 kHz to 20 kHz, is a major concern to automotive manufacturers because of noise, vibration and harshness performance and warranty-claim related costs. The prediction of brake squeal is as difficult as ever due to nonlinearities, uncertainty boundary and operating conditions. The exact contact conditions between pad linings with rotor are often not known and difficult to model. The friction contact has been found to be multi-scaled, inhomogeneous, and highly dependent on operating conditions. In this study, the surface roughness of a pad lining is measured using a Nanovea profilometer with a P1 – OP3500 measurement pen. Statistical distributions are generated for describing the surface roughness. A FE full brake model is updated by modal testing of individual components, subassemblies and full assembled brake. The variation in roughness of pad lining is implemented into the FE model for analysing its effect on brake instability prediction using the conventional complex eigenvalue analysis. The instability predictions are compared with noise dynamometer tests. Results are discussed with a view to applying the methodology to quantify the reliability of squeal prediction against the uncertainty in contact conditions.

Structural Acoustics and Vibration (others):

Paper ICA2016-602

Experimental characterization of dry friction isolators for shock vibration isolation

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Abstract

An overview of the use of shock isolators based on dry friction is briefly presented and the possibilities for the development of a more efficient shock isolation system are discussed. Cable isolators, also known as wire rope are studied. Such isolators present nonlinear stiffness behaviour in different directions, i.e. tension-compression, roll and shear, as well as dry friction damping, and are known for being excellent shock isolators. However, little is known about the actual dynamic behaviour under shock loading. Different commercially available samples are studied for several configurations and load rates. The advantages of the use of cable isolator over a classic linear system with viscous damping is also discussed. The stiffness and damping of these models are quantified experimentally, and then this data is used to validate the estimated shock response. The combination of these two properties in a further mathematical model is suggested for the modelling of dry friction isolators available off the shelf.

Acoustic Array System: Near-Field Acoustic Holography and Vibro-Acoustic Field Reconstruction:

Paper ICA2016-489

Utilization of wall reflection data in the acoustic pyrometry for estimating the gas temperature in a duct section

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Abstract

Estimation of temperature field of the medium in a duct section is required for monitoring and controlling the combustion status of various power systems. Acoustic pyrometry is the most promising technique for this purpose. The usual acoustic pyrometry concept is using the measured retarded time data of sound propagation between multiple sets of acoustic sensors and actuators, which are to be used for the inverse calculation employing a proper basis function. A large number of retarded time data are needed for accurate temperature estimation; however, due to practical limitations, the number of data representing the direct propagation path is usually a small number. This study suggests the additional use of the retarded time data from the wall reflection of sound to append to the direct data. Numerical simulation is conducted for a rectangular duct section having predetermined multiple hot spots of the medium. Compared to the results using the conventional pyrometry, a clear improvement in finding the position of hot spots and estimating the temperature contours can be observed for such a two-dimensional complex temperature field. It is also found that the reconstruction result is better when the separation between temperature hot spots is wide enough.

Acoustic Array System: Near-Field Acoustic Holography and Vibro-Acoustic Field Reconstruction:

Paper ICA2016-501

Wideband acoustical holography

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Abstract

Near-field Acoustical Holography methods are limited to relatively low frequencies where the average array inter-element spacing is less than half a wavelength, while beamforming provides useful resolution only at medium-to-high frequencies. With adequate array design, both methods can be used with the same array. But for holography to provide good low-frequency resolution, a small measurement distance must be used, while beamforming requires a larger distance to limit side lobe issues. Wideband Holography was developed to overcome that practical conflict. Only a single measurement is needed at a relatively short distance and a single result is obtained covering the full frequency range. The rather new Compressive Sensing methods have started making it possible to use irregular array geometries for holography up to frequencies where the average array inter-element spacing is significantly larger than half of the wavelength. In general, these methods allow reconstruction of a signal from sparse irregular samples under the condition that the signal can be (approximately) represented by a sparse subset of expansion functions in some domain, i.e., with the expansion coefficients (amplitudes) of most functions equal to zero. The underlying problem solved is that at high frequencies the microphone spacing is too large to meet the spatial sampling criterion, and thus there is no unique reconstruction of the sound field. A reconstruction must therefore have a built-in “preference” for specific forms of the sound field, thus a smooth form of the reconstructed sound field is enforced. The present paper describes a new method called Wideband Holography (WBH),

which is covered by a pending patent. WBH uses a dedicated iterative solver that enforces the required sparsity.

INVITED

Acoustic Array System: Near-Field Acoustic Holography and Vibro-Acoustic Field Reconstruction:

Paper ICA2016-567

Near-field acoustic imaging based on Laplacian sparsity

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Abstract

We present a sound source identification method for near-field acoustic imaging of extended sources. The methodology is based on a wave superposition method (or equivalent source method) that promotes solutions with sparse higher order spatial derivatives. Instead of promoting direct sparsity, as in standard compressive sensing or basis pursuit approaches, solutions with a piecewise constant gradient or curvature are promoted, suitable for modeling extended sources that are subject to smooth spatial variations. The obtained results are compared to Least Squares and Compressive Sensing solutions, and the validity of the wave extrapolation used for the reconstruction is examined. It is shown that this methodology can overcome conventional limits of spatial sampling, and is therefore valid for wide-band acoustic imaging of extended sources.

INVITED

Acoustic Array System: Near-Field Acoustic Holography and Vibro-Acoustic Field Reconstruction:

Paper ICA2016-603

Measurement of a target's structural admittance and the prediction of its scattered field in different media

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Abstract

This research presents a new method to measure the scattered pressure field from a resonant target, based on a noise-based measurement of the surface structural admittance matrix Y_s. This matrix characterizes the elastic properties of the target and relates the total normal velocity v to the total pressure p (both spanning the total surface) at the target surface, that is, $v = Y_s p$. We show that Y_s can be constructed through cross correlations of these velocity and pressure measurements when the target is excited by a fully diffuse field, viz. an incoherent superposition of randomly phased plane waves in all directions. The surface velocity and pressure measurements represent two holograms, and it is well known that two holograms are necessary to separate the incident and scattered fields from one another. This separation provides the ingredients to reconstruct the structural admittance matrix. Once we reconstruct Y_s, the bistatic scattered field can be calculated for any incident direction. Furthermore, we show that one can accurately predict the scattering when the target is placed in a different medium (proud, half-buried) without redoing the experiment. The theory is verified by a numerical experiment on a target consisting of a thick homogeneous, air-filled sphere. The diffuse field is created by a pulsed loudspeaker source moved to 300 positions around the sphere. Microphones

and surface accelerometers are used to produce holograms of the surface pressure and velocity over a frequency band of 1 kHz - 6 kHz. The *in vacuo* admittance Y_s is accurately determined as well as the bistatic scattering signature to a plane wave at arbitrary incidence with the external medium water.

Acoustic Array System: Near-Field Acoustic Holography and Vibro-Acoustic Field Reconstruction:

[Paper ICA2016-607]

Localization of a defect in an impacted plate using time-domain nearfield acoustical holography and time evolution of spatial features

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Abstract
In this proceeding, a method for localizing defects in a structure by analysing its radiated sound field is presented. The studied structure is a Plexiglas plate with a hole of 4.5 cm in diameter. The plate is impacted at its centre. The radiated pressure field is sampled using a microphone array located in the nearfield of the source. A formulation for time-domain nearfield acoustical holography developed by the authors is used to calculate the pressure field at the source. An important feature of this formulation is that it considers three-dimensional linear convolution to avoid wrap-around errors. The determination of the back-propagated pressure field is an ill-posed problem, and regularization is performed using Tikhonov's method and generalized cross-validation. An image processing algorithm is then applied to the back-propagated sound field to localize the defect. This algorithm assumes that the defect is small compared to the wave front and that it produces a discontinuity in the impedance of the structure. To pinpoint possible defect positions, the algorithm calculates and compares statistical features between pixels at the same radius from the impact point. Reflection of the sound pressure at the pinpointed locations is then used to confirm the actual position of the defect.

Tuesday afternoon, 6 September 2016 **Room 204**
16:30 - 16:50
Signal Processing in Acoustics
SP2 - Acoustic Array System: Near-Field Acoustic Holography and Vibro-Acoustic Field Reconstruction

INVITED

Acoustic Array System: Near-Field Acoustic Holography and Vibro-Acoustic Field Reconstruction:

[Paper ICA2016-813]

Separation of track contribution to pass-by noise by near-field array techniques

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Abstract
A technique to separate the track noise contribution is proposed based on identifying and extracting the track signature from the pass-by noise information measured with a microphone

array relatively close to the track. Separation of the contributions of the vehicle and the track in the pass-by noise spectra is a challenging task, which is currently addressed by a combination of direct and indirect measurements and model predictions. Due to the uncertainties in the separation of the track contribution, whether a vehicle will comply with regulations during certification tests is still very much track dependent. Therefore, accurate means to identify the track contribution to the pass-by noise are needed. In this paper we propose to make use of the fact that in a wide frequency range the track is a distributed source that radiates plane waves at a given angle with respect to the track. By measuring the sound field close to the track with a microphone array, the wavenumber spectrum of the radiated sound can be determined. For the track contribution this wavenumber spectrum is tonal and therefore sparse. We make use of this property to design filters that extract the track contribution to the total pass-by noise. This is illustrated with simulations and experiments.

Tuesday afternoon, 6 September 2016 **Room 204**
16:50 - 17:30
Signal Processing in Acoustics
SP3 - Model-Based Optimization/Estimation and Analysis

Model-Based Optimization/Estimation and Analysis:
Paper ICA2016-272

Nonlinear signal processing techniques for active sonar localization in the shallow ocean with significant environmental uncertainty and reverberation

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Abstract
Sonar signal processing techniques based on acoustic models of shallow ocean environments are frequently of limited use for the mid- to high-frequency regimes typical for active sonar. To make use of acoustical models of the environment, signal processing algorithms typically require better-than-a-wavelength accuracy in the acoustic path estimates. Given this limitation, and practical knowledge that can be expected for shallow ocean environments, model-based signal processing schemes are often limited to frequencies below approximately 1 kHz. Recently, a new nonlinear signal processing technique (see Worthmann et al., JASA 138, 3549-3562, 2015) was able to localize mid-water-column, high frequency sources in the shallow ocean despite imperfect knowledge of the acoustic environment. The new technique takes advantage of a nonlinear construction called the autoproduct to controllably create difference frequencies to recover out-of-band, low frequency field information from in-band, high frequency hydrophone measurements. This passive source localization technique is extended to monostatic active sonar target localization, where strongly reverberant environments can obscure a desired target echo. The frequency difference active sonar technique is presented along with comparisons to existing detection and localization algorithms. Additionally, simulations are provided of these algorithms’ performance in a 200-m deep ideal waveguide with strong reverberation and environmental uncertainties that includes a mid-water-column target at 5-km range, using broadcast frequencies between 2 kHz and 5 kHz. Successful detection and localization of this target using this nonlinear frequency difference scheme is found to be possible at signal-to-reverberation levels as low as –12 dB in this simulation.

Model-Based Optimization/Estimation and Analysis:

Paper ICA2016-141

Methodology for the study of array processing employed for the analysis of source directivity

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Abstract

Knowledge of the acoustic source radiation pattern is essential in sound rendering. Hence, in recent years there has been a growing interest in studies of acoustic radiation patterns of real sources such as musical instruments. Extensive recordings of musical instruments were performed recently with spherical microphone arrays. Due to practical constraints, these spherical arrays were composed of a small number of microphones that cannot capture the entire spatial information of the directivity pattern, especially at high frequencies. Hence, algorithms that can effectively model the radiation of an acoustic source sampled by a finite number of microphones have been studied recently. In this paper, measures for the performance of such algorithms are developed, based on errors due to finite spherical harmonics order, spatial aliasing, and error due to the axis-symmetry assumption. Then, the performance of the axis symmetric projection algorithm applied to COMSOL simulations of a loudspeaker source was analyzed. The results of this analysis suggest that this algorithm has the potential for improving the modeling accuracy of the radiation pattern.

Tuesday afternoon, 6 September 2016 **Room 204**

17:30 - 18:50

Signal Processing in Acoustics

SP4 - Signal Processing in Acoustics (others)

Signal Processing in Acoustics (others):

Paper ICA2016-111

About Doppler-Fizeau effect on radiated noise from a rotating source in cavitation tunnel

R. Boucheron

DGA Techniques hydrodynamiques, Val-de-Reuil, France, romuald.boucheron@intradef.gouv.fr

Abstract

The operational requirements for naval and research vessels have seen an increasing demand for quieter ships either to comply the ship operational requirements or to minimize the influence of shipping noise on marine life. To estimate the future radiated noise of a ship, scale measurements are realized in tunnel. DGA Hydrodynamics owns its cavitation tunnel with low background noise which allows such measurements. Among all noise origins, the present paper considers the problem of a rotating blade. Due to flow and pressure load of each blade faces, blades vibrate and ring out at modal resonance frequencies which is a strategic problem for acoustic discretion. The frequency emitted by the blade will be measured by the non-moving hydrophone with a variable shift in frequency due to Doppler effect. The first part of the paper deals with the idealized case of a monopole in rotation for which theoretical equations for the spectra are derived. The classical Bessel-shape spectra obtained for a sine-shape modulation is modified due to non-stationary modulation, imposed by the time dependence of Mach number contribution in Doppler equation. We have developed an algorithm to compute its associated spectra dealing with Bessel function properties. The second part of the paper will consider the more realistic case of a dipole source in tunnel. The algorithm has been adapted to take into account the dipole directivity. A short parametric study on different parameters as rotation velocity, hydrophone position and dipole orientation concludes the proposed communication.

Signal Processing in Acoustics (others):

Paper ICA2016-728

Perceptual audio coding schemes based on adaptive signal processing tools

Fernando A. Marengo Rodriguez[a], **Sergio A. Castells**[b], **Gonzalo D. Sad**[c]
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Abstract

In this paper, new perceptual audio coding schemes based on adaptive processing tools are proposed. They rely on both the empirical mode decomposition (EMD) and the ensemble empirical mode decomposition (EEMD). In comparison with other perceptual coding schemes, the one presented here is simpler since physically meaningful components of the input signal are detected , then their extrema are extracted and Golomb-Rice encoding of the extracted samples is performed. The proposed scheme is assessed in terms of compression ratio and perceptual quality for various tracks from the EBU SQAM CD. The obtained results are compared with those corresponding to other perceptual audio coding methodologies.

Signal Processing in Acoustics (others):

Paper ICA2016-608

Effects of acoustic degradations on cover song recognition

Julien Osmalskyj[a], **Jean-Jacques Embrechts**[b]
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Abstract

Cover song identification systems deal with the problem of identifying different versions of an audio query in a reference database. Such systems involve the computation of pairwise similarity scores between a query and all the tracks of a database. The usual way of evaluating such systems is to use a set of audio queries, extract features from them, and compare them to other tracks in the database to report diverse statistics. Databases in such research are usually designed in a controlled environment, with relatively clean audio signals. However, in real life conditions, audio signals can be seriously modified due to acoustic degradations. For example, depending on the context, audio can be modified by room reverberation, or by added hands clapping noise in a live concert, etc. In this paper, we study how environmental audio degradations affect the performance of several state-of-the-art cover song identification systems. In particular, we study how reverberation, ambient noise and distortion affect the performance of the systems. We further investigate the effect of recording or playing music through a smartphone for music recognition. To achieve this, we use an audio degradation toolbox to degrade the set of queries to be evaluated. We propose a comparison of the performance achieved with cover song identification systems based on several harmonic and timbre features under ideal and noisy conditions. We demonstrate that the performance depends strongly on the degradation method applied to the source, and we quantify the performance using multiple statistics.

Signal Processing in Acoustics (others):

Paper ICA2016-581

Robust acoustics and speech perception of aerial robot under ego noise for scene understanding

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Abstract

For aerial robot based robust scene understanding, it is required that each sensor on the aerial robot acquire reliable sensor data. Due to the flight condition of the aerial robot and the ensuing emergency situation mission, however, the robot platform suffers a large uncertainty of data consistency and differences with normal ground condition. Information bearing acoustic sources such as human speech and other acoustic events representing abnormal situation particularly represents such case as the airborne platform encounters noise bearing sources convoluted with wind noise. This paper addresses these problems and develops signal processing approaches for acoustic noise reduction and signal enhancement under harsh environments such as ego/wind/propeller fan noise of aerial robot so that speech event can be extracted for time critical scene (especially calling for help) understanding. In particular, we explore advanced acoustic signal processing techniques to identify the relevant approaches including cancellation of platform generated ego-noise and fan noise which are convoluted by wind noise during flight. In addition, this paper develops effective sensor fusion arrangements for realizing adaptive directivity by combining the advantages of microphone array so that sporadic speech source or speech events can be captured under environmentally adverse conditions.

Tuesday afternoon, 6 September 2016 **Auditorium 2**
14:30 - 16:10
Communication Acoustics
CA2 - Communication Acoustics

INVITED
Communication Acoustics:
Paper ICA2016-187

The advent of communication acoustics in retrospect

Jens Blauert
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Abstract
Communication Acoustics is a cover label for those aspects of acoustics that involve relations between the classical fields of acoustics and the information and communication technologies. The usage of the term started around 1974, but it took 42 year until it finally became an explicit topic at the International Congress of Acoustics, namely, here in Buenos Aires at the ICA 2016. In the current talk, the history of Communication Acoustics will be recalled, considering the roles of electro-acoustics, auditory perception and audio-signal processing in the course of the development of this field. In this context, two areas of application will be taken as examples to discuss the essence of Communication Acoustics, namely, (a) Virtual-Reality (VR) generation and (b) Computational Auditory-Scene Analysis (CASA) ─ both dealing with parametric representations of auditory scenes. In both of these fields the trend can identified of including more explicit knowledge as well as learning algorithms into Communication-Acoustics systems and their components. For this purpose, proficiency in computational symbol processing is required in terms of scientific craftsmanship, besides pure signal-processing skills.

INVITED
Communication Acoustics:
Paper ICA2016-790

Hearing research relevant to communication acoustics

Barbara Shinn-Cunningham
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Abstract
Hearing research has been a foundation for understanding communication acoustics. Historically, hearing researchers have viewed individual differences as a nuisance that makes it difficult to interpret how acoustic conditions affect auditory perception. This paper sketches out how we have begun to use individual differences to tease apart the processes that affect perception in young adults who have normal hearing thresholds, with a particular focus on how listeners understand speech when there are competing sound sources. We find that individual subjects show consistent differences in their ability to understand speech in noise, which are correlated with differences in the ability to extract fine temporal details of sounds, as well as with physiological differences in the fidelity with which the brainstem encodes temporal acoustic detail. Growing evidence suggests that cochlear synaptopathy,

otherwise known as "hidden hearing loss," may explain these differences in otherwise healthy listeners with normal hearing thresholds. After reviewing this evidence, this talk will consider the implications for developing new technologies that could assist listeners who have difficulty communicating in noisy, reverberant settings.

INVITED

Communication Acoustics:

Paper ICA2016-109

Evolution of sound reproduction – from mechanical solutions to digital techniques optimized for human hearing

Ville Pulkki

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Abstract

Sound reproduction consists of the processes of recording, processing, storing and recreating sound, typically speech, music, environmental or other sounds. The applications include such fields as public address, telecommunication, music, cinema, virtual reality, and aided hearing. The unifying factor is the common endpoint of the chain, the human listener. Historically, reproduction of sound has come a long way from the first monophonic phonographs. Nowadays audio is available efficiently in various digital formats and immersive 3D spatial sound can be reproduced over different multichannel set-ups or headphones. In general view, the trend in the development of sound reproduction during last decades has been the dedicated design of reproduction systems to match better the resolution of human hearing system. The reproduction methods are designed to reproduce acoustic parameters in time, frequency, and space with only a bit better resolution than the hearing system has. This paper discusses the needs and challenges faced in sound reproduction and it will also present various solutions and technologies.

INVITED

Communication Acoustics:

Paper ICA2016-391

Views on sound quality

Alexander Raake

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Abstract

The talk presents an overview of research on Sound Quality evaluation. In related research, sound quality refers to the judgment of auditory events with regard to internal references. Sound quality is closely related with Quality of Experience (QoE), that is, quality judgments addressing auditory experiences. The talk reviews and discusses different research streams regarding "sound quality" and QoE, including work on product sounds, audio and speech technology. The analysis addresses assessment using human listeners as measurement entity as well as approaches involving quality models, that is, algorithms that target predictions of human judgments. Perspectives for future research are provided, as well as considerations on limitations of practical assessment strategies.

INVITED

Communication Acoustics:

Paper ICA2016-496

On the contribution of spatial hearing to speech intelligibility

Steven van de Par[a], Sarinah Sutojo[a], Esther Schoenmaker[a]

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Abstract

A spatial separation between a target speaker and interfering sources is known to improve speech intelligibility. Various effects may contribute to this: the target-speech-to-interferer ratio may be better

in one ear, binaural unmasking may improve the detectability of target speech components, and spatial cues may facilitate better segregation of the target speech from the background. An experiment will be discussed that uses a stimulus manipulation that reduces the possibility of having a binaural unmasking effect. Interestingly, these manipulations have little effect on speech intelligibility and this small effect can be explained based on the percentage of glimpses that is available in the stimulus. In a second experiment, the stimuli are manipulated in such a way that only the most salient proportion of the target speech glimpses is located at a different position than the interfering speech sources. This very small amount of spatial cues is already enough to create a strong spatial advantage in speech intelligibility. The presence of the sparse spatial cues seems to create a more effective processing of monaural cues in line with ideas of auditory grouping and stream segregation.

Tuesday afternoon, 6 September 2016 **Auditorium 2**
16:30 - 18:10
Communication Acoustics
CA2 - Communication Acoustics

INVITED

Communication Acoustics:

Paper ICA2016-465

Concert halls – conveyors of musical expressions

Tapio Lokki
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Abstract

The first question of an acoustician regarding a new concert hall is too often: What is the reverberation time? Such tradition has historical reasons as reverberation time and sound strength have been considered the most important objective measures of concert halls. However, although they are indeed important—music should be loud enough and reverberated to sound good—they do not tell much on the multidimensional acoustical conditions of a hall. This paper collects the recent research to understand concert hall acoustics beyond these traditional measures. The results indicate that a concert hall is considered exceptional if it renders music expressive, i.e., conveys the subtle nuances of music from the stage to the audience as well as possible. In other words, a concert hall is preferred if it renders proximate sound with large dynamics and wide frequency range. This paper links together musical acoustics, room acoustics and human spatial hearing to highlight recent findings in concert hall acoustics research.

INVITED

Communication Acoustics:

Paper ICA2016-318

Modern advancements in audiology

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Abstract

Population estimates suggest that the number of people aged over 60 years will increase from 605 million to 2 billion between the years 2000 and 2050. Age-related hearing loss or presbycusis progresses with age, resulting in significantly handicapping hearing loss for 80% of people over 80 years of age. It has become increasingly clear that the long-term effects of hearing loss extend beyond the peripheral reduction of audibility. A ground swell of concern has built around plausible cognitive influences of untreated hearing loss. For this reason, a scientific eye has turned toward the prospective inclusion of cognitive screening at the time of audiologic assessment: a behavior that has been most common among physicians, nurses, and psychologists. Data on the relationships between cognitive ability, hearing status, and the treatment of hearing loss will be reviewed. Technical advancements in auditory prostheses have allowed for great strides in the treatment of hearing loss. Bi-directional transmission of broadband audio and data between ear-worn devices (whether they be hearing aids or cochlear implants) is a reality that, among other acoustic signal processing, allows for binaural directional beamforming, addressing the common complaint of speech understanding in

noise. Similar advances have introduced low-power audio and data transmission directly from hearing aids to mobile telephones. This connection between medical and mobile devices is allowing for a spate of novel developments that have demonstrated meaningful benefits for people with hearing loss.

Communication Acoustics:

Paper ICA2016-43

A binaural model to segregate sound sources in the presence of early reflections using a multi-source precedence-effect model

Jonas Braasch[a], **Nikhil Desphande**[b]
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Abstract

In this paper a binaural model is presented that can segregate two spatialized speech signals in the presence of early reflections and late reverberation. For each sound source, the model identifies the lateral positions for the direct sound component and an early reflection using a precedence effect model. The model then uses a filter to eliminate the reflection of the sound source it wants to extract. Afterwards the equalization/cancellation (EC) method is used to select those time/frequency bins where the binaural cues correspond to the localization cues of the desired sound source. It is shown that the model is sufficiently robust to deal with late reverberation, and it is also shown that the model performs better when the reflections are removed prior to the EC analysis.

Communication Acoustics:

Paper ICA2016-818

Source-blind binaural source segregation utilizing head movement

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Abstract

This model takes a mixture of two simultaneous speech signals, spatialized with head-related transfer functions to unique azimuth positions, and extracts either source using the equalization/- cancellation (EC) method. The model localizes the sources by analyzing interaural correlation and then virtually rotates its head to find the orientation for the best resulting signal-to-noise ratio. Next, the model segments the mixed speech signal in time and frequency bins, and uses an EC algorithm in each bin to compensate the target signal from the mixture. From the residual noncancelled energy, it generates a binary map and overlays this on the spectrogram. The ability of the model to cancel out the target signal determines the bins where the target is actually present. The signal is then reconstructed in time and frequency, leaving only one desired target signal. Resulting signal-to-noise ratios after cancellation consistently approach 80 dB.

Communication Acoustics:

Paper ICA2016-317

Experiment on listening difficulty of announcements from municipal public address system by applying the method of continuous judgment by category.

Junichi Mori[a], **Syun Sonoda**[b, c], **Fumiaki Satoh**[c] **and Hideki Tachibana**[d]
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[d] Professor Emeritus, The University of Tokyo, Japan, pon-t@iis.u-tokyo.ac.jp

Abstract

Municipal public address (M.P.A.) system for disaster prevention is widely used in Japan. Speech intelligibility of M.P.A. announcements, however, tends to be deteriorated by multi-pass echoes with long time delay owing to the sounds from the loudspeaker systems covering other subareas and the

reflections from nearby buildings. After the Great East Japan Earthquake in 2011, the importance of M.P.A. systems has been realized again and various studies are being made for the design of the M.P.A. system. The authors also have been investigating this research topic by putting emphasis on speech intelligibility of the M.P.A. announcements by performing auditory experiments. In those experiments, listening difficulty of the announcements was judged by overall impression after hearing a test sound. The subjective impression of listening difficulty, however, changes every moment according to the extent of overlap of the announcements caused by the multi-pass echoes. To examine such instantaneous auditory impression, the method of continuous judgment by category developed by Namba et al. was applied in this study. As a result, it has been suggested that the method of continuous judgment by category is effective to examine the instantaneous listening difficulty and to compose the M.P.A. announcements with robustness against the effects of multi-pass echoes.

Tuesday afternoon, 6 September 2016 **Auditorium 3**
14:30 - 16:10
Physical Acoustics
PA3 - Ultrasound

INVITED

Ultrasound:

Paper ICA2016-543

Effect of convective drying assisted by ultrasound on drying time and aroma of tamarillo (*Cyphomandra betacea* Cav. *Sendt*) and mango (*Mangifera indica* L.) fruits

Kamila Méndez[a], Natalia Salazar [a], Juan Ocampo[a], Diana Manrique[a], Catalina Álvarez[a], Carlos Orrego[a]
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Abstract
New trends in global healthy food consumption have increase the production of fresh fruits and fruit based value-added products as a result of the awareness of their high contents of bio-active compounds like carotenoids, vitamins, minerals, dietary fiber and antioxidants. High moisture content in fruits can generate high losses in post-harvest handing, storage and distribution. Dehydration process based on water activity reduction, is a common option for overcoming such losses. Regular hot air drying could affect negatively the quality properties of the fruit due to the long residence time at high temperature. Power ultrasound (US) application during the convective drying has been used as a new method able to decrease drying time. The aim of this work was to evaluate the effect of the convective drying assisted by ultrasound on drying time and aroma losses of tamarillo slices (TS) and mango slices (MS). An experimental design was developed for both fruits, showing the positive effect of ultrasound during convective drying process. Power ultrasound (20 kHz, 45 W) was applied in different cycles during drying test at 50°C, applying 5 minutes and 10 minutes of ultrasound each half hour intermittently. Weigh loss of samples was measured every hour until reaching moisture content of 10.0±1.0% (wet basis). Aroma losses measurement was development by SPME method, and the results were compared with the fresh fruit data for all the drying tests.

Ultrasound:

[Paper ICA2016-48]

Converting sunlight into audible sound: Some practical measurements on the Heliophone

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Abstract

This paper presents measurement results obtained with a device called Heliophone. The Heliophone collects and converts sunlight to sound without electronic amplification. Sun light is focused to a photoacoustic piston surface in the photoacoustic cavity by means of a compound parabolic collimator, and its intensity is modulated by a mechanical chopper. The photoacoustic cell is connected to an acoustic horn, which acts as an impedance matching device between the cavity and the open air environment, making the sound audible. Acoustic measurements are presented, using sun light to drive the Heliophone.

Ultrasound:

[Paper ICA2016-595]

Particle-image-velocimetry investigation of flow morphology at sudden expansion and contraction in oscillating flows

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Abstract

Concerns about viscous dissipation losses at the inlet and exist of stacks in thermoacoustic devices have encouraged investigations of flow morphology around sudden expansions and contractions under oscillating flow conditions. In this work, a set of experiments were carried-out to investigate the flow morphology at the edges of a set of parallel plates placed inside a thermoacoustic resonator. The effects of different plate thicknesses (4 mm and 7 mm), plate separations (3 mm and 6 mm) and drive ratios (0.76 % and 1.82 %) were studied. Particle image velocimetry was used to capture the instantaneous velocity fields for the different cases allowing visualization and quantification of the flow morphology, size of the vortex core and size of the disturbance zone for each case. The measured free-stream velocities were compared to simulations made by DeltaEC and the observed velocity fields were compared to numerical simulations made by computational flow dynamics and reasonable agreements were found. Results indicate that increasing the plate thickness (corresponding to using thicker stack walls in thermoacoustic devices) projects the formation of larger vortices at the edge of the plate and increases the size of disturbance zone around the plate and decreases the vortex-vortex interaction along the wall of the same plate. Decreasing the plate spacing (corresponding to using denser stacks in thermoacoustic devices) has the effect of increasing the flow disturbance forcing the vortex structures generated closer to each other thus forming more complex flow structures with larger

disturbance zones. Increasing the drive ratio (corresponding to higher loading of the thermoacoustic devices) results in a larger vortex core and in a larger size of the disturbance zone.

Ultrasound:

[Paper ICA2016-26]

High resolution imaging in challenging media: a model-based approach

Ludovic Moreau[a], **Eric Larose**[a]

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Abstract

In nondestructive evaluation, most ultrasonic imaging methods are based on deterministic approaches where the ultrasound energy is focused in the propagation medium to identify a reflector. Several focusing strategies have been tested, such as time-reversal mirrors, beamforming algorithms, or a combination of both such as the CAPON and MUSIC algorithms. These methods are robust to the noise level. However, they suffer inherent weaknesses that limit the range of possible applications. For example, as soon as the travelling paths of ultrasound cannot be identified accurately, artefacts appear in the images. Typically, this happens when reflections and/or mode conversions occur in the medium. In multiple scattering media, such methods can therefore not be used. Moreover, the imaging resolution of these methods is limited to objects which size is of the order of the wavelength. We introduce an imaging method where the measured data are reconstructed with a numerical model. The approach consists in describing the medium as an ensemble of elastic and geometric parameters, and to infer these parameters via an efficient inversion algorithm. This approach allows super-resolution imaging capabilities with calculation times of the order of a few minutes. Numerical and experimental application are presented, including the imaging of complex defects with guided waves and the imaging of cracks in a multiple scattering medium (concrete).

Ultrasound:

[Paper ICA2016-29]

Light diffraction by large amplitude ultrasonic waves in liquids

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Abstract

Light diffraction from ultrasound, which can be used to investigate nonlinear acoustic phenomena in liquids, is reported for wave amplitudes larger than that typically reported in the literature. Large amplitude waves result in waveform distortion due to the nonlinearity of the medium that generates harmonics and produces asymmetries in the light diffraction pattern. For standing waves with amplitudes above a threshold value, subharmonics are generated in addition to the harmonics and produce additional diffraction orders of the incident light. With increasing drive amplitude above the threshold a cascade of period-doubling subharmonics are generated, terminating in a region characterized by a random, incoherent (chaotic) diffraction pattern. To explain the experimental results a toy model is introduced, which is derived from traveling wave solutions of the nonlinear wave equation corresponding to the fundamental and second harmonic standing waves. The toy model reduces the nonlinear partial differential equation to a mathematically more tractable nonlinear ordinary differential equation. The model predicts the experimentally observed cascade of period-doubling subharmonics terminating in chaos that occurs with increasing drive amplitudes above the threshold value. The calculated threshold amplitude is consistent with the value estimated from the experimental data.

Ultrasound:
[Paper ICA2016-94]

Experimental and numerical results on the evaluation of the adhesion level in a bonded tri-layer metal/adhesive/metal, using lamb guided waves

Camille Gauthier[a;b], Mihai Predoi[c], Damien Leduc[a], Mounsif Ech-Cherif El-Kettani[a], Jocelyne Galy[b]
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[b]IMP UMR CNRS 5223, France
[c]University Politechnica of Bucharest, Roumania

Abstract
The aim of this work is the evaluation of the quality of the adhesion in a tri-layer Aluminum/Epoxy/Aluminum. The quality of the adhesion is progressively enhanced by applying different surface treatment on the substrate surface, and considering different cure of the epoxy layer. This work is collaboration between physical chemists for the samples manufacturing and acousticians for ultrasound characterization, and an extension of previous studies on bilayers Aluminum/Epoxy adhesion evaluation. The goal is to discriminate acoustically each level of adhesion using guided Lamb waves. The experimental study is conducted using a contact transducer as an emitter and a laser vibrometer as a receiver to measure the time dependent normal displacement of the propagating waves, at the aluminum surface. This allows to obtain the experimental dispersion curves. A rheological Jones model which replaces each interface substrate/adhesive by springs distribution is solved by a semi analytical finite element (SAFE) method. The determined dispersion curves, depending on the springs stiffnesses introduced in the model, are then compared to the experimental ones.

Ultrasound:
[Paper ICA2016-103]

Extension of theoretical acoustics to curvilinear spacetime

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Abstract
Einstein's nonlinear field equations of general relativity show that energy and momentum produce curvature of spacetime and gravitational force is a fictitious force due to the curvature of spacetime. The flat spacetime is only an approximation of the curvilinear space-time. Usually in theoretical acoustics, the flat spacetime is used. The gravitational force will be of significance when the energy and momentum is substantial. This will be the case of nonlinear acoustics such as dealing with the acoustic radiation force which is intense sound which can be illustrated by the force of levitation which is acoustic radiation force and here the gravitational force will also play a part in the acoustic levitation. Another example is cavitation when the heat energy release is enormous of the order of fusion energy. In this paper, we will first explain the meaning of general relativity and followed by the application of the Einstein field equations to cavitation and force of levitation and the influence of large electromagnetic fields on the response of quartz crystal resonators. The action principle will be used. This will enable a more accurate treatment of certain topics in theoretical acoustics and bring the field to the next level.

Ultrasound:

[Paper ICA2016-174]

Evolution modelling of fast and slow magnetoacoustic waves in thermally unstable plasma

Dmitrii Zavershinskii[a,b], Nonna Molevich[a,b], Igor Zavershinskii[a], Sergei Pichugin[b] Dmitrii Ryashchikov [a, b]

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Abstract

The nonlinear evolution of fast and slow magnetoacoustic waves in the plasma medium with non-adiabatic heating/cooling processes is under consideration. We assume that the magnetic field vector is inclined at an arbitrary angle to the direction of magnetoacoustic wave propagation. The non-adiabatic processes depend on temperature and density and result in the steady non-equilibrium state of the medium. The steady state caused by the balance between heating and cooling rates gives possibility for various thermal instabilities to appear. In current paper, we discuss the wave mode of thermal instability (so–called isentropic instability) and neglect the presence of other modes of thermal instability. The isentropic mode influences on acoustic and magnetoacoustic waves and causes wave amplification. The linear analysis predicts simultaneous amplification of fast and slow magnetoacoustic waves with different increments. Furthermore, our analysis predicts simultaneous disintegration of fast and slow waves on the sequences of autowave (self-sustaining) shock pulses. These results are proved by the numerical simulation of full system of one-dimensional magneto-hydrodynamic equations. The simulation is conducted using the implicit fully conservative difference scheme. The results of numerical modeling show the disintegration of initial perturbation on the sequence of fast and slow shock pulses. The parameters of obtained autowave pulses are in good agreement with values predicted by our analytical model.

Ultrasound:

[Paper ICA2016-175]

Acoustically induced transitions of CHFD discharge in swirl flow

A.O. Gorbunova[a,b], N.E. Molevich[a,b], A.I. Klimov[c] , S.S. Sugak[a], I.P. Zavershinskii[a], D.I. Zavershinskii[a,b]

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Abstract

A mechanism providing acoustically induced generation of helical disturbances on a plasma filament in swirling flows is presented. The numerical simulation shows a formation of a helical structure of the gas flow in the presence of an acoustic standing wave. The flow structure determines the heat flux. In the heated zones, the ionization rate increases and the channel resistance drops that leads to the formation of a new discharge channel, which takes the helical shape of the flow structure. The shape of the simulated helical structure can be matched with the experimentally observed large-amplitude disturbances of the plasma filament in the swirling flow with an excited standing acoustic wave.

Ultrasound:

[Paper ICA2016-251]

Overturning of nonlinear acoustic waves in media with power-law attenuation

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Abstract

Without incorporation of weak-shock theory, the lossless Burgers equation predicts a multivalued waveform for nonlinear propagation beyond a certain distance. Inclusion of thermoviscous attenuation, which increases quadratically with frequency, prevents the occurrence of multivalued waveforms. The

same is true for any attenuation law that is proportional to frequency raised to an exponent greater than unity. For exponents less than unity the situation is less clear. For example, when attenuation is constant with frequency (exponent equal zero) there is a critical value of the source amplitude below which a multivalued waveform is predicted and above which it is not. Prediction of multivalued waveforms indicates that the mathematical model is inadequate and must be either supplemented by weak-shock theory, or augmented to include an additional loss factor. To investigate the prediction of multivalued waveforms subject to power-law attenuation with exponents between zero and unity a Burgers equation with the loss term expressed as a fractional derivative is used [Prieur and Holm, J. Acoust. Soc. Am. 130, 1125–1132 (2011)]. Transformation of the equation into intrinsic coordinates following Hammerton and Crighton [J. Fluid Mech. 252, 585–599 (1993)] permits simulation of waveforms beyond the point at which they become multivalued. These solutions are used to determine the parameter space in which initially sinusoidal plane waves are predicted to evolve into multivalued waveforms for power-law attenuation with exponents less than unity.

Ultrasound:

Paper ICA2016-264

Time-resolved imaging of electrically excited GHz acoustic vibrations at arbitrary frequencies

Osamu Matsuda,[(a)] **Hirofumi Shono,**[(a)] **Shun Kato,**[(a)] **Shogo Kaneko,**[(a)] **Sylvain Mezil,**[(a)] **Motonobu Tomoda,**[(a)] **Oliver Wright**[(a)]

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Abstract

Time-resolved two-dimensional imaging of vibrations can be achieved using the optical pumpprobe technique: ultra-short light pulses (pump pulses) with sub-picosecond temporal width are focused onto a sample to impulsively generate acoustic vibrations, and the generated acoustic displacement is detected by delayed light pulses (probe pulses) with an optical interferometer. By scanning the pump-probe delay time and the probe spot position, one can obtain the spatiotemporal evolution of the acoustic field. The technique has previously been applied to isotropic/ anisotropic media, microstructures and phononic crystals to reveal their acoustic properties such as the dispersion relation. Though the technique is quite powerful, it is sometimes required, in particular in device applications, to excite the acoustic vibrations electrically rather than with optical pulses. In this paper, we show how to image GHz acoustic vibrations generated electrically through the use of inter-digital piezoelectric transducers working at an arbitrary frequency. The detection is done optically using the above mentioned two-dimensional imaging technique. The use of a lock-in amplifier with an appropriate reference signal allows one to retrieve the amplitude and phase of the acoustic vibration at any point in the image. This techniques should allow the extension of two-dimensional GHz acoustic imaging to more practical applications.

Ultrasound:

Paper ICA2016-296

Ultrasonic fatigue test and Comsol finite element modelling

Juan Carricondo[(a), (b)]**, Martín Iofrida**[(a), (b)]**, Augusto Bonelli Toro**[(a)]**, Guido Ferrari**[(c)]**, Martín Gómez**[(b), (c)]

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Abstract

This work on ultrasonic fatigue in carbon steels is framed in the Plan Argentina Innovadora 2020 national science policy. Theoretical foundations of ultrasonic fatigue test were studied, linking the operation principles of a commercial equipment and using the finite element modelling as a comparison method. A carbon steel alloy was tested, defining an appropriate geometry for the test sample based on the ultrasonic equipment requirements and turning process. An offset high voltage divider was designed and fabricated to characterize the equipment used, allowing to measure the signal at the output of the power amplification stage. In turn, finite element modelling of the transducer

and the fatigue specimen were performed by Comsol 5.0 software. The resonance frequency for the system (transducer and fatigue specimen) was calculated obtaining a value of 19592 Hz, close to the 19633 Hz measured with the high voltage divider. Two carbon steel ultrasonic fatigue specimens of the same geometry were tested. The first trial lasted approximately two hours, interrupted when leaving resonance condition as indicated by an equipment built-in led. Non-destructive testing of dye penetrant was performed to locate the crack. However, it was determined that the first specimen did not present any crack. Therefore, a second ultrasonic fatigue test of about eight hours with a new specimen was performed. After the test, the second specimen was fractured when applying a slightly force. Finally, crack was located by SEM microscopy determining an internal crack nucleation at specimen notch, in agreement with studies on the subject.

Tuesday afternoon, 6 September 2016 **Lounge Lateral Room**
15:10 - 15:30
POSTER SESSION - Monitor 1
Architectural Acoustics - Room and Building Acoustics
AA2 - Acoustics of Worship Spaces

POSTER
Acoustics of Worship Spaces:
Paper ICA2016-815

Mexico City’s cathedral: An archaeoacoustical and musicological analysis

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Abstract
In this paper, we discuss the need to study colonial architectural spaces and their relationship with aspects of acoustic design. Other research groups have addressed issues such as the importance of acoustics in the architectural design of churches; how much the composers of a certain era took into account the acoustic of the spaces and venues where their compositions would be played; and how intelligible were the texts of complex polyphonic compositions of the time in certain enclosures, etc. In order to address these questions, they have used mathematical models and digital audio engineering to reconstruct the acoustics of these spaces and give tentative answers to the above questions. Our research group has also considered other topics such as the possible reconstruction of sonorities corresponding to polychoral musical practices, and the Iberian and Latin American musical characteristics of the colonial era. Initially we propose the study of the Cathedral of Mexico City and we present the results of impulse-response analysis of the building and describe our methodological framework and progress made so far.

POSTER

Soundscape, Psychoacoustics and Urban Environment:

Paper ICA2016-329

Expanding sonic imagination in design practices with architectural soundwalks

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Abstract

We sought to expand the horizons of architectural practice to include sonic imagination in design phases, and to investigate how people with different experience in spatial sound listening describe a soundscape and understand acoustic phenomena. Twenty-two film sound design students, took part in a two hour research soundwalk. They were given instructions to listen to the sonic environment around them, represented on a map with a 3d view of the area. They stopped at 6 locations among 30 on the map for a longer time, 4 minutes. Binaural field recordings with orientation data were made for future reference and a sound artist recorded his perspective with another set of microphones. The students were divided into listening (L) and listening and writing (LW) participants. The LW group for each location listened and described sonic textures as sources or sounding words, placing them at the center of a polar diagram. They also noted the acoustics of the places on the 3d diagrams. The LW group followed some exercises, while the L group just listened. During group discussion both groups answered questions on these locations and participated verbally with comments on the experience. Both groups matched the location numbers among the 30 places to questions asking for specific sonic and architectural features. Analysis of the data, reflections on the activity and the benefits of writing when listening for educational and awareness purposes are evaluated and presented herein.

POSTER

Soundscape, Psychoacoustics and Urban Environment:

Paper ICA2016-476

Soundscapes in three areas of theTucumán province, Argentina

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Abstract

This work aims the acoustic adaptation of 3 geographical areas of the province of Tucumán. The objectives of this study are to identify and analyze noise pollution and make recommendations to the community and municipalities and relevant institutions, implementing a methodological strategy of participatory action research. The results achieved are: the study of various points of noise; the quantitative evaluation of them and the way in which this affects people; and, the development of a series of recommendations for users, municipalities and institutions; the generation of material for awareness of the problem; the dissemination and transfer of results. It is concluded that: to) 2 areas of the province of Tucumán: San Miguel de Tucumán and Yerba Buena are under a clear acoustic pollution; and (b) the third area, the commune of San Pablo is characterized for having less actual pollution but a future similar to the previous two. The lack of knowledge and awareness in the society

makes urgent the need to create awareness in the 2 first mentioned areas and the social and cultural revaluation of the third, avoiding in this way reach the levels achieved in San Miguel de Tucumán and Yerba Buena.

Tuesday afternoon, 6 September 2016 **Lounge Lateral Room**
16:30 - 16:50
POSTER SESSION - Monitor 1
Environmental Acoustics & Community Noise
EN1 - Noise Assessment and Control

POSTER

Noise Assessment and Control:

Paper ICA2016-229

Urban acoustic assessment: Urban noise characterization of Miguel Rossafa Square in Umuarama Paraná

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Abstract

Noise pollution is one of the evils that affect urban centers, which main source of noise is vehicular traffic. Despite the fact that the noise is an pollutant agent invisible to the eye, its effects are noticeable in the body and are responsible for reduced life quality. Because of its harmfulness, this topic has been the subject of research aimed at describing the effects of noise, characterize the levels and the relationship between the noise and the environment. This work has the objective to characterize the urban noise of the Miguel Rossafa Square in Umuarama Parana. Through monitoring and analysis of acoustic indices. Ascertain whether they agree with the NBR10151 / 2000 and the criteria evaluation of the United States Department of Housing and Urban Development (HUD). The monitoring was conducted in 2013 and 2014. In each monitoring were measured 5 different points of the square, one being the control point, there were three measurements a day, three days a week, with a duration of fifteen minutes in each point. The average sound pressure levels and statistical levels achieved in 2013 and 2014 exceeded 70 dB (A), especially the L10, which represents the peak values, showed 83.9 dB (A) in 2013, and 83.62 dB (A) in 2014. The measured values were shown to be higher than 60 dB (A), maximum allowed by the standard. The research has shown that noise sources, especially the traffic of vehicles and surrounding commercial activities, altered the quality of the sound environment, in reason that the levels are above the permissible by the current regulations. Moreover, it proved to be essential to adopt measures and a urban and environmental planning in order to minimize noise levels and therefore the effects on individuals who attend the analyzed area.

Tuesday afternoon, 6 September 2016 | Lounge Lateral Room
16:50 - 17:10
POSTER SESSION - Monitor 1
Environmental Acoustics & Community Noise
EN2 - Noise Mapping

POSTER
Noise Mapping:
[Paper ICA2016-478]

The place of the tram regarding to the 49/2002/EC Directives Noise Map

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Abstract
The Environmental Noise Directive (END) 49/2002/EC strategic noise mapping regulation describes the representation of noise pollution coming from the tramline, as well as its treatment as a part of the railway noise map. Experience shows that the locating of the tramlines is foreign on the railway noise maps. We present the results of several studies, all of which demonstrate that the tramline is to be treated as part of public road transportation. In particular, this fact is reflected in the importance of the preparation of the action plan. The planning of noise reduction for city roads can't be made without that the noise pollution coming from the public road, as well from the tram line is not represented together on the noise map. We illustrate with specific Hungarian and Romanian examples (Hungary treats together the tram and public road transportation, while Romania in accordance with the END directive represents the noise pollution coming from the tramlines together with the railway noise pollution) the noise maps and action plans produced through two different interpretation. We present the advantages of the Hungarian method, which primarily prevails at the preparation of action plans.

Tuesday afternoon, 6 September 2016 | Lounge Lateral Room
17:10 - 17:30
POSTER SESSION - Monitor 1
Environmental Acoustics & Community Noise
EN3 - Road Traffic Noise Modeling and Noise Barrier

POSTER
Road Traffic Noise Modeling and Noise Barrier:
[Paper ICA2016-321]

Comparison of the insertion loss of noise barriers with different shapes of upper structure

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Abstract
Noise barriers are widely used in and around the residential area. However, the noise reduction efficiency has not been satisfied with the comfort of residents. In order to increase the noise reduction of the noise barriers, various shapes of upper structures have been introduced and used nowadays. The present study investigates the insertion losses of the noise barriers with different shapes of upper structure. Seven different upper structures were used and computer simulations were undertaken to calculate the insertion losses of noise barriers under same conditions. As a result, it was found that insertion loss of noise barriers with Y and T-shaped upper structures is greater than those with and L-shaped upper structures. It is because the Y and T-shaped upper structure overgrow along the horizontal axis of the noise sources and measurement points while and L-shaped upper structures stretch to one side of noise barrier. Also, 1/2 scaled mock-up models were made with Y-shaped and 1-shaped upper structure. Acoustic measurements were undertaken at the different positions away from the barriers. Through the experiments, it was found that insertion loss of noise barriers with Y-shaped upper structures is greater than those 1-shaped upper structures in most frequencies. The difference

of insertion loss was most great (5.8dB) at the point 1m away from the barrier and it decrease with the increasing distance from the barrier up to 1.7dB which is 8m away from the barrier.

Tuesday afternoon, 6 September 2016 **Lounge Lateral Room**
15:10 - 16:10
POSTER SESSION - Monitor 2
Biomedical Acoustics
BA1 - Biomedical Acoustics

POSTER
Biomedical Acoustics:
Paper ICA2016-292

Tympanic membrane physiology

Santos Tieso[a], Lucas Fantini[a], Francisco Messina[a], Nahuel Cacavelos[a], Gilda Farelli[a], Leonardo Zavala[a], Maria Tieso[a], Sebastian Iezzi[a], Federico Bosio[a]
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Abstract
The surface of the tympanic membrane defines the amount of acoustic energy received and transferred to the ossicles and was taken as a parameter of analysis in the study of the middle ear impedances. Low sound pressure stimuli creates in the middle ear the need to transfer the entire content of energy incident on the outer surface of the tympanic membrane to the cochlea. For this to happen, the middle ear must maximize the effective pressure on the oval window, in respect of which affects the tympanic membrane. However, it is not convenient to transfer all incident energy in the tympanic membrane to the inner ear when levels of sound pressure are high, since they are harmful to hair cells. In this paper physical model is developed to explain the energy transfer mechanism of the tympanic membrane.

POSTER
Biomedical Acoustics:
Paper ICA2016-324

Detection of abnormal lung sounds considering spectral and temporal features of heart sounds

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Abstract
In this paper, we propose a robust classification method for lung sounds contaminated with heart sounds in order to distinguish between healthy subjects and abnormal patients with pulmonary emphysema. We previously developed a classification procedure based on a maximum-likelihood approach by using hidden Markov models (HMMs). However, contaminated heart sounds caused difficulties in achieving a highly accurate classification, because it was difficult to generate HMMs that distinguished between adventitious sounds and heart sounds with high accuracy, by using power and spectral acoustic features only. To address this problem, we propose a classification technique that is based on the use of spectral features and temporal features related to heart sounds: distributions of durations and time intervals of heart (S1) sounds. A validity score for detected adventitious sounds and heart sounds in the classification process is designed by considering the distribution of time intervals of heart sounds and differences in the durations between the adventitious sounds and the heart sounds. In the proposed method, the total likelihood of each respiratory sound is obtained by summing the spectral likelihood derived from the HMMs and the validity score. In the classification of healthy subjects and patients using 94 lung sound samples from 94 subjects, the proposed method achieved a higher classification rate (90%) than the baseline method (84%) using only the spectral features, thus demonstrating the superiority of the proposed method.

POSTER

Biomedical Acoustics:

Paper ICA2016-499

Continuous wavelet transform for tissue periodicity estimation: effect of noise and scatterers position variability

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Abstract

Quantitative ultrasound (US) provides quantitative data in an attempt to overcome the high subjectivity of ultrasonography. Scatterers spacings (SS) are one of the parameters investigated. The continuous wavelet transform (CWT) has been considered for a more accurate search of signal singularities. The aim of this work was to evaluate the effect of signal noise level and scatterers positions variability (jitter) on periodicity characterization using CWT. US signals were simulated with mean scatterer spacings (*MSS*) of 1.0, 0.99, 0.9, 0.8, 0.7, and 0.6 mm as well as with several *jitters* (variation in the expected scatterer position – 0 to 30%), and noise as defined by *Ad* (percentage ratio between the average echo amplitude from diffuse to regular scatterers – 1 to 50). The identification of the modulus maxima sequence was done after applying CWT. SS distributions were computed for the estimations of MSS average and standard deviation (in mm), variation coefficient (*VC*), average histogram mode (in mm) and accuracy error (*e%*). Kolmogorov-Smirnov tests were applied to verify if distributions between simulated sets were from the same population ($\alpha = 0.05$). For $Ad \leq 30$, CWT was able to estimate *MSS* with e% of ± 2%, and it could discriminate differences of 0.1 mm ($p < 0.001$) even at the maximum jitter and *Ad* simulated. A greater *VC* was observed for high noise levels (*VC* up to 14.74%), even with no jitter (*VC* = 12.76%). Histogram modes tend to decrease with increasing noise. These results corroborate other works in indicating that the use of MSS alone may not discriminate different periodical media. CWT may be an option for mapping SS distributions. Spectral analysis associated with CWT can be a choice in the future to improve the performance of this quantitative approach.

Tuesday afternoon, 6 September 2016 | **Lounge Lateral Room**

16:30 - 17:30

POSTER SESSION - Monitor 2

Noise: Sources and Control

NS4 - Materials for Noise Control

POSTER

Materials for Noise Control:

Paper ICA2016-604

Calculation of noise emitted by technology equipment soundproofing compartment

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Abstract

There is a large number of noisy equipment requiring arrangement of soundproof enclosures to limit the spread of noise in modern civil and industrial buildings. Noise reduction is due to the noise source enclosure designs with high sound insulation. At that a part of the sound energy penetrates through the insulating enclosure housing, a variety of leaks and holes and is radiated into the surrounding air space. Thus housing becomes a source of noise. When selecting and designing the soundproof

housing assemblages one must carry out assessment of the sound energy spread from them as from secondary sources of noise. The method of calculation of the direct sound pressure levels emitted by the covers is considered in the paper, taking into account the characteristics that affect the distribution of the sound energy in the environment. The easier casing shape as compared to equipment form, stability and predictability of the noise emission factors of its elements allow to apply more accurate methods for calculating the levels of direct sound in its near field. For the calculation of the direct sound from the housing it is proposed to use the integral expression, allowing determining the amount of sound energy density at the reference points as the result of summing the contributions from each segment of housing surface. The technique of this calculation is described and the accuracy of the proposed method is estimated by comparing the calculation results with the experimental data.

POSTER

Materials for Noise Control:

Paper ICA2016-660

Membrane acoustic metamaterial with multiple magnetic negative stiffness cells

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Abstract

A membrane absorber with single magnetic negative stiffness cell usually can achieve a thin layer sound absorption for one absorption peak at low-frequency. In this paper a membrane acoustic metamaterial with multiple magnetic negative stiffness cells was designed to realize a compact design for low frequency sound absorption. This absorber comprises three cells in which exists different magnetic negative stiffness. And the electro-acoustic analogy model of this absorber for three cells face the incident wave in parallel is present to calculate the total impedance and predict its absorption performance. The multiple frequencies sound absorber is capable of unity absorption at multiple frequencies and broad bandwidth, due to the regime of different magnetic negative stiffness. This has been well demonstrated by equivalent circuit simulation and experiment results in this paper.

POSTER

Materials for Noise Control:

Paper ICA2016-701

Review of the acoustic property of the open graded friction course (OGFC)

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Abstract

The road noise, resulting from the accumulation of noise emissions of vehicles, currently constitutes a serious problem of environmental quality. In order to expand the studies on the pavement using a coating with OGFC, in southern Brazil, it is aimed to check the acoustics functional properties of the layer, wich were performed by laboratory test of five type of OGFC mixtures and compared with two conventional blends of asphalt concrete (AC) - (range B and C). Then, was performed in the laboratory the characterization of materials for molding of the test specimens following the Marshall methodology of asphalt mixtures and selection of grain sizes. The asphalt cements oil (CAP) used were binder CAP 60/85 and CAP 55/75. The sound absorption tests were conducted according to the procedure specified in ISO 10534-2:1998, and was performed in a impedance tube at ambient temperature. Also, a metal mold was used at the end of the tube to represent a rigid termination, where the sample was inserted. So, were generated graphics of sound absorption coefficient versus frequency domain for each tested specimen. The noise reduction coefficient (NRC) was calculated for each sample too, following the ASTM C 423. Thus, the OGFC samples showed major mean absorption than AC samples, where 5 types of OGFC had their NRC results between 0.25 and 0.35. In other hand, the dense mixed sample (range B and C) gave a NRC result of 0.20. Therefore, is proved that the use of porous asphalt mixtures as coating has demonstrated some advantages with regard to reduction of noise generated by the tire-pavement interaction.

Tuesday afternoon, 6 September 2016 **Lounge Lateral Room**
15:30 - 15:50
POSTER SESSION - Monitor 3
Acoustical Measurements and Instrumentation
SI1 - Sound Intensity and Inverse Methods in Acoustics

POSTER
Sound Intensity and Inverse Methods in Acoustics:
Paper ICA2016-533

Inverse acoustic characterization of rigid frame porous materials from impedance tube measurements

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Abstract
We will present a method for the inverse characterization of rigid frame porous materials using audible frequency acoustic measurements in an impedance tube 3 cm in diameter. We recover the six acoustical parameters of the Johnson-Lafarge model, namely porosity, tortuosity, viscous and thermal characteristic lengths and flow and thermal resistivities. The proposed method is based on a minimization process, where the quantities of interest are found as the minimizing values for the difference between measured and modeled density and compressibility. A scattering matrix formulation is used to obtain the reflection and transmission coefficients R and T, found as the elements of the scattering matrix, and to obtain the effective density and compressibility in the range of 200 - 6500 Hz. Five different porous materials with flow resistivity ranging from 2,000 to 60,000 Ns/m4 are tested and the results of the inversion process are compared to direct measurements of the acoustical quantities as well as to an already established recovery method developed by Olny and Panneton. The results are also validated against measurements with a rigid backing. It is found that the proposed method can be used to recover the material parameters quickly and reliably.

Tuesday afternoon, 6 September 2016 **Lounge Lateral Room**
15:50 - 16:10
POSTER SESSION - Monitor 3
Acoustical Measurements and Instrumentation
SI2 - Acoustical measurements and instrumentation

POSTER
Acoustical Measurements and Instrumentation:
Paper ICA2016-784

Calibration of piezoelectric accelerometers at INTI

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Abstract
The National Institute of Industrial Technology, INTI, is the technical referring of the argentinian state in the assistance for the industrial development in the country. Besides, is the National Institute of Metrology, (NMI) due to its legal responsibility on the national measurement standards. The SI units of physical quantities, such as meter per squared second (m/s^2) for the quantity of acceleration, is realized in the Vibrations Laboratory, UT Acoustics of INTI ,and disseminated to external customers through calibrations of transducers and equipment. In former times, INTI participated in the first

intercomparison in acceleration that took place in America within the SIM (Interamerican System of Metrology) with the participation of the NMIs of Canadá, EEUU and México (NORAMET) and Brasil and Argentina (SURAMET). The frequency range comprised 50 Hz to 5000 Hz. The system for the primary calibration of accelerometers is a simple Michelson laser interferometer with a single detector and an electro-dynamic shaker according to the guidelines given in ISO16063-11. In the present work, recent improvements in the method are described, which allowed to extend the frequency range from 10 Hz to 10000 Hz. The charge sensitivity results of a laboratory standard accelerometer B&K 8305 calibration are given, and a good correspondence with the values stated by INMETRO in a previous report was observed. Accordingly, a bilateral SIM comparison will be encouraged in order to validate the current CMCs of INTI.

Tuesday afternoon, 6 September 2016 **Lounge Lateral Room**
16:30 - 17:30
POSTER SESSION - Monitor 3
Electroacoustics and Audio Engineering
EL1 - Electroacoustics and Audio Engineering

POSTER

Electroacoustics and Audio Engineering:

Paper ICA2016-747

Estimation of the Thiele-Small parameters in mobile application using as link Pure Data programming and Microsoft Visual Studio

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Abstract
This paper is about a multiplatform mobile application developed to calculate the Thiele-Small parameters of a loudspeaker using Pure Data. Pure Data is a programming language mainly used with computers, which is known for its versatility in creating electronic music and simulations of wave oscillators. In order to develop this multiplatform mobile application, it has been proposed a computer platform which can be compatible with Pure Data functionalities. In that sense, Microsoft Visual Studio has been chosen and, more concretely, the framework known as Apache Cordova. Pure Data was used to make the Thiele-Small parameters calculations by introducing mathematical formulas and Apache Cordova was used to show the results through the graphical interface of the mobile device. This procedure was performed using a file named WebPd, that works as a library code for compatibility between Pure Data and Apache Cordova. WebPd can be found on the official website of Pure Data library developers. Although there are already several mobile applications created using Pure Data, they have the limitation of being compatible with Apple devices only. So basically, this work was able to use a programming language for computers in the design of mobile device applications. In addition, it is capable to run in almost every operating system used by Smartphones (Android, IOS and Windows Phone). On the other hand, the Thiele-Small parameters mobile application can be considered a useful tool for the design of speaker boxes within academic purposes.

POSTER

Electroacoustics and Audio Engineering:

[Paper ICA2016-425]

Multi-zone sound field reproduction method including distance effects

Taku Shimizu[a;b], Jorge Trevino[a], Shuichi Sakamoto[a], Yôiti Suzuki[a], Tomohiko Ise[c]
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Abstract

Sound field reproduction systems are promising to present spatial sound in ultra-realistic multimedia system. However, most sound field reproduction systems are limited only to single-user applications because re-creating sound fields over extended regions that accommodate multiple listeners requires an unpractical large number of loudspeakers. The problem can be overcome by a multi-zone approach to sound field reproduction, controlling the sound field only in the neighborhood of the listeners. An advantageous property of these systems is their ability to present different sounds to each of their users. Nevertheless, this may also lead to unnatural sound fields that are extremely hard to reproduce. The present research avoids this possibility by introducing a novel sound field reproduction method based on spherical harmonic analysis. The proposal considers a single, shared sound field for all listeners. The target field for each listener is derived from considerations of natural sound propagation. This method has previously been applied to reproduce plane wave sound fields with good angular accuracy inside multiple regions. However, the plane-wave assumption is only accurate when presenting sound sources that are located far away from the listeners. We now extend our previous method to allow for the presentation of point source sound fields across multiple listening regions. To achieve this, we advance a sound field processing stage in the spherical harmonic domain that accounts for the sound source distance effects. The proposal is evaluated through a numerical experiment. Our results show that the proposal can re-create the point-source field accurately in two separate listening regions, including angular parallax between regions and distance-related attenuation.

POSTER

Electroacoustics and Audio Engineering:

[Paper ICA2016-488]

Multiband distortion pedal

Alejandro Serrano
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Abstract

This research study focused on creating a stomp-box consistent with the physiology of the human auditory system, and in this case, to include the tonal characteristics of the electric guitar. Through an applied deductive method, psychoacoustic parameters and circuits of existing distortion pedals were selected; which together with a study of the acoustic and electro-acoustic characteristics of the electric guitar, justified the design of a multiband distortion pedal. To define and design the acoustic capabilities of the pedal, an inductive method was used to select three virtuoso guitarists and four emblematic albums. Subsequently, by way of trial and error the circuit was tested, assembled, and edited. Finally, ten guitarists with different styles and skill levels tested the stomp-box prototype. It was interesting to discover that the pedals that were chosen for this study were the same ones used by the guitarists selected for this study. The multiband process successfully met the psychoacoustics and technical requirements defined. The pedal expanded its versatility to at least five distinct musical styles with up to 15 different configurations. This proves that it is feasible to design devices at a competitive price that reconcile both technical and aesthetic enhancements. The criteria tested in this research study have renewed existing techniques for recording and mixing electric guitar sounds.

Wednesday, 7 September 2016

Wednesday morning, 7 September 2016 **Juan Pablo II Auditorium**
09:00 - 10:40
Environmental Acoustics & Community Noise
EN5 - Wind Farm Noise

INVITED
Wind Farm Noise:
Paper ICA2016-245

Environmental noise due to large wind turbines: what we have learnt

Alice Elizabeth González[a], Pablo Gianoli Kovar[a], Matteo Deambrosi Papini[a], Matías Hernández Castellani[a], Martín Paz Urban[a]
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Abstract
Uruguay began to change its energy matrix some years ago. In that time, the preferred prediction method in the country in environmental impact studies of stationary noise sources was the ISO Standard 9613-2. The Environmental Engineering Department of the Faculty of Engineering (UdelaR) was asked to deeply study the environmental acoustic impacts from large wind turbines and to select an accurate prediction method to be used in the environmental studies to get the environmental license for new wind farms. Thus, we have begun to walk along a wide and complex way. The National Energy Directorate has supported these first studies. We have still a lot to learn but we feel we have technically grown along this 5-years' experience. We have studied about low frequency noise and its adverse human health consequences, the adverse consequences on birds and bats, the different siting criteria and security distances, the compatibility of wind energy devices with other uses of land, and we have focused on the generation and propagation of aerodynamic noise. We have tried some different approaches and finally we arrived to a satisfactory model to describe the main phenomena involved in noise generation: the incoming wind turbulence, the trailing edge noise related to the boundary layer release, and the blade passage ahead the tower. We have achieved accurate results for noise propagation considering geometric divergence and atmospheric absorption for distances greater than 300 m, the minimum security distance from the tower. We have found that the geometric divergence -or the atmospheric refraction of sound pressure waves - is neither squared nor linear, but it is frequency-related.

Wind Farm Noise:
Paper ICA2016-70

Determination and assessment of noise from wind turbines in Germany

Andrea Bauerdorff [a], Thomas Myck [b]
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Abstract
In 2014, 28.2 % of the total electrical energy has been generated from renewable energy sources within the European Union. In this context, wind turbines are of great importance. Especially in Germany, the number of these installations has continuously increased in the last years. Therefore, it is particularly important to have clear legal regulations for noise protection by wind turbines. Wind turbines with a height of more than 50 m are subject to licensing pursuant to the German Federal Immission Control Act. Assessment of noise immissions from wind turbines is carried out according to a General Administrative Provision to the Federal Immission Control Act, which is called "Technical Instructions on Noise Abatement - TA Noise". It describes the methods for the determination and the assessment of noise caused by industrial or commercial installations, including wind turbines. These regulations will be explained in detail. Moreover, the low-frequency noise immissions of wind turbines will be discussed and evaluated.

Wind Farm Noise:

[Paper ICA2016-223]

Risk assessment related to noiseand infrasonic noise in a wind farm

Dariusz Pleban[a], Jan Radosz[b], Bozena Smagowska[c]

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Abstract

The development of wind energy in Poland is accompanied by an increase in the number of people employed in the workplaces of wind energy sector. According to recent estimates, 8,400 persons were employed in wind energy sector, with 600 of them directly employed on wind farms. On the basis of an analysis of the activities carried out by the wind farm staff and interviews with the employees, a typical weekly profile of work including 15 activities was developed. Thereafter, noise and infrasonic noise measurements at workplaces in the wind farm were carried out during service and maintenance activities carried out by the staff. The scope of the measurements at workplaces included the determination of the following parameters: the A-weighted sound pressure levels, the A-weighted maximum sound pressure levels, the C-weighted peak sound pressure levels and the G-weighted sound pressure levels. The results of the measurements made it possible to perform risk assessments related to noise and infrasonic noise. It was concluded that noise and infrasonic noise at the workplaces in the wind farm do not exceed the values of Maximum Admissible Intensities for noise in the working environment and the infrasonic noise nuisance criteria, respectively. The results of risk assessment showed that risk related to exposure to noise and infrasonic noise in the wind farm is small (admissible).

Wind Farm Noise:

[Paper ICA2016-86]

A noise generation and propagation model for large wind farms

Franck Bertagnolio

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Abstract

A wind turbine noise calculation model is combined with a ray tracing method in order to estimate wind farm noise in its surrounding assuming an arbitrary topography. The wind turbine noise model is used to generate noise spectra for which each turbine is approximated as a point source. However, the detailed three-dimensional directivity features are taken into account for the further calculation of noise propagation over the surrounding terrain. An arbitrary number of turbines constituting a wind farm can be spatially distributed. The noise from each individual turbine is propagated into the far-field using the ray tracing method. These results are added up assuming the noise from each turbine is uncorrelated. The methodology permits to estimate a wind farm noise map over the surrounding terrain in a reasonable amount of computational time on a personal computer.

Wind Farm Noise:

[Paper ICA2016-168]

Prediction of environmental sound pressure levels due to large wind turbines

Matteo Deambrosi Papini[a], Matías Hernández Castellani[b], Alice Elizabeth González[c], José Cataldo[d]

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Abstract

In the last years, Uruguay has strongly built wind power in its energy mix. Even if it is intended to be a “green energy”, wind turbines can generate some adverse impacts on the environment. One of them is the related noise emissions. The aerodynamic wind turbine noise is generated as a result of three processes

that make the pressure field to fluctuate: 1) Turbulence of wind, which is due to pressure fluctuations around the blades, is variable over time. It is called "incoming edge noise". 2) Release of vortexes from boundary layers developed on solid surfaces of the wind turbine, such as blades, tower and nacelle due to the viscous forces. It is intended to be a continuous noise that is called "trailing edge noise" 3) The passage of the blades ahead of the tower, which imposes a fluctuation of the levels of noise emitted by the abovementioned phenomena. It results in an amplitude-modulated noise. It is called "blade passage noise". The incoming edge noise is caused by the fluctuation of pressure along the blade produced by the incoming turbulence. It can be described by combining the turbulence spectrum modelled by the approach of Von Karman and the aerodynamics theory. The trailing edge noise was determined by applying the turbulence spectrum from Von Karman to the boundary layer thickness, thus determining the pressure drop in the main vortexes. For a typical three-blade wind turbine, the noise immission level in any down-wind site can be intended as the result of the contributions of these three phenomena all along each blade. This theoretical model has shown accurately predictions of the sound pressure levels at distances between 300 m and 1000 m away from the machines.

Wednesday morning, 7 September 2016 **Juan Pablo II Auditorium**
11:00 - 12:00
Environmental Acoustics & Community Noise
EN5 - Wind Farm Noise

INVITED
Wind Farm Noise:
Paper ICA2016-239

Wind turbine noise prediction

Ramani Ramakrishnan[(a)], Vipul Sehrawat[(b)]
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Abstract
Renewable energy sources, however green they may be, have inherent negative concerns. One such source, a multi-bladed wind turbine, generates noise levels that can adversely affect residences that surround a given wind farm. Regulatory agencies require wind farm developers to submit an acoustic assessment report before approving the wind farm. The main focus of the assessment is to evaluate the potential noise levels that can exist at all nearby receptors. Earlier studies applied different prediction models to evaluate the receptor noise levels. However, all of the current available methods assume the wind turbine to be a point source located at the hub height. But, the wind turbine is a multi-bladed rotating source interacting with the mast during its rotation. The main aim of the current investigation is to research the point source assumption. A small 5 turbine wind farm in southwest Ontario, Canada will be used for the study. A few locations, including some very close to two turbines, will be used to conduct acoustic measurements. Three different prediction models will be used to evaluate the noise levels at the different locations. Comparison of prediction and site measurements will shed light on the point source approximation. The results of the study are presented in this paper.

Wind Farm Noise:
Paper ICA2016-353

Propagation phenomena associated with noise due to the operation of large-sized wind turbines

Matías Hernández[(a)], Martín Paz Urban[(a)], Matteo Deambrosi Papini[(a)], Alice Elizabeth González[(a)]
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Abstract
Noise pollution due to the operation of large-sized wind turbines is a current important issue in Uruguay: there is about 1000 MW of installed capacity of wind energy generation in the country. This work was developed in the Department of Environmental Engineering IMFIA of the Faculty of Engineering of the Universidad de la República (UdelaR), Uruguay, in the framework of a project

funded by the National Agency for Research and Innovation (ANII-FSE 10942). The phenomena of generation of aerodynamic noise in these energy generation devices have been presented in detail in another paper of our team. Here, phenomena related to the propagation of aerodynamic noise of large-sized wind turbines are theoretically analysed. A propagation model in free atmosphere, considering the phenomena of geometrical divergence and atmospheric absorption is proposed. The prediction of sound pressure levels is performed by third octave bands. The values of atmospheric absorption coefficients are obtained according to ISO 9613-1 standard for all frequencies of interest. The adjustment is focused on the exponent of the depletion law, which is not square neither linear nor homogeneous in the different third-octave bands considered. The model has been validated with empirical data for distances greater than 300 m from the source. A good fit for different atmospheric stability conditions was achieved.

Wind Farm Noise:

[Paper ICA2016-673]

Sound propagation from a wind turbine in a hilly environment

Timothy Van Renterghem

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Abstract

Sound propagation in the atmospheric boundary layer can be strongly affected by vertical gradients in the wind speed and air temperature, and by atmospheric turbulence. In addition, undulating terrain will either partly shield or focus sound waves, highly impacting sound exposure levels. These aspects become especially important in case of wind turbines present on a ridge, such placement often being efficient to harvest wind energy due to the wind speeding up. In this study, sound propagating from a wind turbine, positioned at a ridge, towards receivers along a valley, is numerically simulated with the Parabolic Equation (PE) method. A combination of an analytical starting field and the conformal mapping method shows to be a useful and numerically efficient approach, overcoming some issues as identified and discussed in this work. Example calculations show the importance of the valley ground impedance for wind turbine noise exposure.

Wednesday morning, 7 September 2016 — **Dr. Valsecchi Auditorium**

09:00 - 10:40

Signal Processing in Acoustics

SP1 - Acoustic Array Systems

INVITED

Acoustic Array Systems:

[Paper ICA2016-731]

Spherical harmonic smoothing for DOA estimation of fully coherent sources

Byeongho Jo[(a)], Jung-Woo Choi[(b)]

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Abstract

The subspace-based estimation of direction of arrivals (DOAs) of multiple sound sources requires multiple observation data from a microphone array, such that basis of signal subspace can be fully identified. For coherent sources, however, multiple measurements only produce linearly dependent observations, which makes it difficult to separate signal subspace from noise subspace. In order to obtain linearly independent observations from fully coherent sources, various smoothing techniques including spatial, temporal, and frequency smoothing have been proposed. In principle, smoothing techniques acquire linearly independent observations by separating a single measurement into multiple subarray data. For constructing subarrays, however, several constraints on the array shape or waveform are need to be satisfied. In this work, we develop a smoothing technique based on spherical harmonic expansion that can be applied without any constraint on the array shape or waveform. The proposed technique directly produces linearly independent observations from spherical harmonic coefficients of measured data. Spherical harmonic coefficients of measured data are separated into

multiple subsets in spherical harmonics domain, which correspond to linearly independent observations of fully coherent sources. The proposed smoothing technique is processed in the spherical harmonics domain, so it is compatible with various eigenbeam-based approaches such as EB-MUSIC and EB-ESPRIT, as far as spherical harmonic coefficients can be measured up to a finite order greater than one.

INVITED

Acoustic Array Systems:

Paper ICA2016-158

Experimental investigation of multiple-input multiple-output systems for sound-field analysis

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Abstract

Spherical microphone and loudspeaker arrays have been widely studied for the acquisition of spatial sound-field information. Recently, a theoretical framework, based on systems that combine both arrays, was presented for the spatial analysis of enclosed sound fields. Such systems are referred to as multiple-input multiple-output (MIMO) systems, and they provide means for an enhanced spatial analysis. However, their performance is limited by errors due to spatial sampling and system model mismatch. The effects of these errors on the system performance were studied recently in theory, without experimental validation. Therefore, the practical usefulness of MIMO systems for room-acoustics analysis has yet to be determined. This paper presents an initial investigation in this direction. MIMO system performance and limitations are first evaluated in a simulation study. The system is then studied experimentally, through the analysis of room impulse responses (RIRs). Experimental validation is achieved in several aspects. First, system properties are studied and compared to previous theoretical results. Then, MIMO processing methods are applied for a spatial analysis of early reflections in the RIR, showing that early room reflections can be identified experimentally. The results of this investigation suggest that MIMO systems can be employed in practice for various applications of room acoustics.

INVITED

Acoustic Array Systems:

Paper ICA2016-204

Frequency dependent coprime linear microphone arrays for direction of arrival estimation using model-based Bayesian inference

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Abstract

Estimating direction of arrival of sound sources is an important acoustical problem often tackled by the use of microphone arrays. Conventional linear microphone arrays have been widely used, provided sufficiently close inter-element spacing for the frequencies of interest and wide enough aperture for beam narrowness. Coprime linear microphone arrays represent an innovative sparse sensing technique which extends the frequency range of a given number of array elements by exceeding the spatial Nyquist limit. Two uniform linear subarrays with equal length and coprime number of elements each operate at wavelengths shorter than half of their inter-element separation, introducing predictable aliasing which only overlaps in one direction, forming a single beam for the overall array. Whereas initial coprime array theory was derived based on an operating frequency based on the specific coprime spacing, recent research shows advantages of broadband beamforming. Parametric models

describing this behavior enable the use of model-based Bayesian inference for estimating not only source direction, but also number of sources present in the sound field, which is often unknown prior to estimation. Nested sampling is used for efficient, sufficiently complete exploration of the likelihood space in the model-selection step to determine number of sources. Parameter estimation then provides a precise estimate of the direction of arrival.

INVITED

Acoustic Array Systems:

Paper ICA2016-312

Flexible microphone array based on multichannel nonnegative matrix factorization and statistical signal estimation

Hiroshi Saruwatari[a], Kazuma Takata[a], Nobutaka Ono[b], Shoji Makino[c]

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Abstract

In this paper, we propose a novel source separation method for the hose-shaped rescue robot based on multichannel nonnegative matrix factorization (MNMF) and statistical speech enhancement. The rescue robot is aimed to detect victims' speech in a disaster area, wearing multiple microphones around the body. Different from the common microphone array, the positions of microphones are unknown, and the conventional beamformer cannot be utilized. In addition, the vibration noise (ego-noise) is generated when the robot moves, yielding the serious contamination in the observed signals. Therefore, it is important to eliminate the ego-noise in this system. Blind source separation is a technique taken to separately estimate the sources without knowing the sensors' positions. Several methods, e.g., independent component analysis, independent vector analysis, and spatially rank-1 MNMF (Rank-1 MNMF) have been proposed so far, but their separation performance is not sufficient. To address this problem, in this study, first, supervised Rank-1 MNMF is proposed, thanks to the stationary properties of the ego-noise, where we train spectral bases of the ego-noise in advance. Secondly, to reduce the mismatch problem between the trained bases and the spectrogram in observed data, we propose an algorithm that an all-pole model is estimated to deform the bases using the reliable spectral components sampled by the statistical signal enhancement method. Thirdly, we propose to initialize Rank-1 MNMF by using the low-rank representation of the estimated speech spectrogram, and improve the convergence. Finally, we reveal that the proposed method outperforms the conventional methods in the source separation accuracy via experiments with actual sounds observed in the rescue robot.

INVITED

Acoustic Array Systems:

Paper ICA2016-412

A parametric superdirective beamformer with uniform linear microphone arrays

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Abstract

Superdirective beamforming has attracted much interest in acoustic, speech and audio processing since it has the potential to achieve the maximum directivity factor (DF) for noise, interference, and reverberation suppression. However, the superdirective beamformer is sensitive to sensors' noise and mismatch between sensors, which considerably restricts its use in practical systems. Therefore, how to achieve a relatively large DF with a reasonable white noise gain (WNG) is becoming an important issue in superdirective beamforming. This paper studies this problem based on the use of a parametric gain as the cost function, which combines the DF and the WNG in one single formula. By maximizing this gain, we derive a parametric superdirective beamformer. Through properly choosing

the parameter order within a small range, this beamformer can achieve a good compromise between a high value of the DF and a low value of the WNG.

Wednesday morning, 7 September 2016 **Dr. Valsecchi Auditorium**
11:00 - 12:00
Signal Processing in Acoustics
SP1 - Acoustic Array Systems

Acoustic Array Systems:
Paper ICA2016-122

Composite aeroacoustic beamforming of an axial fan

Jeoffrey Fischer[a], Con Doolan[b]
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Abstract
As part an effort to more completely understand fan noise, this paper is concerned with the measurement of axial fan noise using beamforming acoustic arrays. Measurements were obtained using a large axial fan rig, located in UNSW's aerospace research laboratory. The rig contains an axial fan, containing 8 rotors and 8 stators. The diameter of the fan is 0.9 m and the rotors are contained in a duct with a flared inlet. The fan operates at 1900 RPM. Further, the input shaft is mounted upstream of the rotor and is housed within a central nacelle. This rather complicated arrangement means that an acoustic array cannot be placed directly upstream of the fan, nor can it be placed parallel to it. Also, the external duct and central nacelle prevent line-of-sight of an array to parts of the fan. To overcome these difficulties, a composite beamforming methodology has been devised. In this method, beamforming images are obtained from two (or more) viewing angles and the sound maps are corrected for to account for the geometrical viewing angle. Two resulting beamforming outputs are then superimposed to obtain composite beamforming sound maps that reveal the sound sources more completely. The results show the various sound sources such as rotor stator interaction and blade self noise as a function of frequency. The influence of the duct and nacelle on the beamforming output was determined by placing an acoustic source on a single blade and measuring the response on the array for multiple angular positions of the fan. Overall, the composite beamforming methodology was found to work well and it is able to overcome the difficulties presented by the axial fan design.

Acoustic Array Systems:
Paper ICA2016-139

Analysis and design of time-domain first-order circular differential microphone arrays

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Abstract
Circular differential microphone arrays (CDMAs) are characterized as compact superdirective beamformers whose beampatterns are almost frequency invariant. In contrast to linear differential microphone arrays (LDMAs) where the optimal steering direction is at the endfire, CDMAs provide almost perfect steering for all azimuthal directions. Herein, we present the design of a first-order CDMA in the time domain which is motivated by several aspects. First, time-domain implementation is important in some applications where minimal delay is required, such as realtime communications. Moreover, direct design in the time domain can reduce the computational efforts compared to the frequency-domain design, especially when short filters are sufficient. We present a design example for the time-domain first-order CDMA illustrating some of its fundamental properties as well as the equivalence to the frequency-domain alternative.

Acoustic Array Systems:
[Paper ICA2016-593]

Towards an enactive robot audition architecture

Valentín Lunati[(a)], Patrick Danès[(b)], Claudia Arias[(c;d)], Fernando Bermejo[(a;d)], Fernando González[(a)], Juan Rosales[(a)], Rodrigo Peréz[(a)]
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Abstract

Robots are usually equipped with advanced capabilities in order to autonomously adapt to real and dynamic environments and to interact with humans. Robot Perception is being inspired by new embodied cognition approaches that redefine the notions of perception, cognition and action, basic processes of intelligent behaviour. Enactive approaches consider the perceptual act as a consequence in action of the structural coupling between the organism and its environment in seek of significance. There is a growing interest in the development of robot perception systems based on new architectures to materialize naturally these action-perception functions. In this direction, we propose an evolution of the EAR sensor [4], which fulfills the constraints of mobile robot audition, such as embeddability, synchronous multichannel acquisition and real-time execution. Using Systems-on-a-Programmable-Chip (SoPCs) methodology, this sensor incorporates a novel architecture that offers all the basic calculation blocks necessary to perform most binaural and array auditory functions, and allows to easily develop new functionalities and connections between motor and perceptual modules in order to implement enactive behaviour. Moreover, Microelectromechanical Systems (MEMS) microphones have been studied and implemented, enabling the acquisition of high-fidelity audio on inexpensive and portable devices. In this paper, performance results are presented for sound source detection and localization functions, and progress is shown towards the implementation of MEMS microphones in Human-Machine Interfaces and Robot Audition. Finally, evolutions towards an interdisciplinary design of enactive audition functions are discussed.

Wednesday morning, 7 September 2016 **Cardenal Pironio Auditorium**
09:00 - 10:40
Architectural Acoustics - Room and Building Acoustics
AA1 - Acoustics in Education

INVITED
Acoustics in Education:
[Paper ICA2016-178]

A comprehensive survey on the current acoustic environment in child day care centers in the Kumamoto city region, southwest Japan

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Abstract

This is a report on survey results gathered from questionnaires and site visits conducted at licensed child day care centers in Kumamoto City, a regional core city of approximately 740,000 population, and its adjacent Koshi City (app. 60,000 population). The survey involved 132 centers, and responses were received from 2,130 childcare providers. Results from the site visits indicated that about half of the classes in the centers were acoustically separated by sound insulators such as fixed walls, while the remaining half utilize movable partitions or multiple classes held in a single room. Only 13% of day care rooms used sound-absorbing materials. Many childcare providers responsible for children aged three or older, who have more conversation than younger children, reported problems involving throat damage and difficulty conducting conversations. There were indications that using sound-absorbing materials in childcare rooms and separating class spaces could help to ameliorate these situations.

INVITED

Acoustics in Education:

Paper ICA2016-241

Effects of aircraft noise on children's reading and quality of instruction in German primary schools: results from the NORAH study

Jan Spilski[(a)], Kirstin Bergström[(a)], Ulrich Möhler[(b)], Jochen Mayerl[(a)], Thomas Lachmann[(a)], Maria Klatte[(a)]

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Abstract

Earlier studies mostly show that chronic exposure to aircraft noise may impair children's cognitive development, especially reading acquisition. In the framework of the NORAH-study, the effects of aircraft noise on reading and verbal precursors of reading acquisition were investigated in 1,090 second-graders from 29 schools in the vicinity of Frankfurt/Main Airport. Aircraft noise levels at schools and at the children´s homes were calculated based on radar data from the Flight Track and Monitoring System (FANOMOS), provided by German Air Traffic Services. Although aircraft noise levels at schools did not exceed 60 dB (LAeq 8-14) and were thus considerably lower when compared to prior studies, multilevel analyses revealed significant effects of aircraft noise on children´s reading comprehension after adjustment for individual (e.g. socioeconomic status) and class-level (e.g., road traffic noise, classroom insulation) factors. A 10 dB increase in aircraft noise was associated with a decrement of one-tenth of an *SD* on the reading test, corresponding to about one month reading delay in this test. Sensitivity analyses confirmed the robustness of the results. Teachers' reports indicate impairments of classroom instruction due to aircraft noise. This relationship could be a possible reason for the mainly negative effects of aircraft noise on reading performance.

INVITED

Acoustics in Education:

Paper ICA2016-524

Teaching in a classroom environment: Speech adjustment to real and simulated classroom conditions

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Abstract

School teachers have an elevated risk of voice problems due to vocal demands in the workplace. Such demands can lead to changes in vocal effort, a term used to describe physiological changes in voice production as vocal loading increases, which can be quantified in terms of dB SPL. This paper summarizes the results of three studies addressing voice use in real and simulated classroom settings. In the first study, 57 teachers were observed over two weeks in teaching and nonteaching environments. Differences were observed between the two environments and across the course of the day. In a second study conducted to better understand how teachers may adjust from one environment to the next, the speech of 20 talkers was recorded in order to evaluate how vocal effort is affected by speaking style and room acoustics. The participants read a text aloud talkers in a semi-reverberant room was recorded in three different styles corresponding to soft, normal and loud levels, both with and without artificial multi-talker child babble, and with and without polycarbonate panels at 1m from the subject. A third study was conducted to investigate the effects of changes in the acoustical environment when these changes were visually concealed. Forty-five participants performed a short vocal task in two different rooms: a variable-acoustic room and an anechoic chamber. Talkers were taken back and forth between the two rooms; each time they entered the variable-acoustic room, the reverberation time and/or the background noise had been modified, although the room was visually the same. Generally, the results of the studies indicated that SPL increased in occupational vs. nonoccupational contexts and in the loud style relative to the normal style. Some vocal behavior differences between male and female talkers were observed.

INVITED

Acoustics in Education:

Paper ICA2016-278

Voice and classroom interactions – recent outcomes

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Abstract

During the last decades, several European countries released acoustical norms, guidelines or recommendations for schools. The main focus in the mentioned documents was on speech intelligibility and on background noise levels necessary for sufficient concentration of students during the educational process. However, overall acoustical comfort in classroom includes also other aspects related to acoustics, such as vocal effort of teachers. Recently, several researchers have shown that teachers belong to the large social group with chronic laryngeal fatigue and the topic became a discussed issue at conferences and acoustic forums. The main research questions have been addressed to issues such as "how acoustic conditions in classrooms affects the vocal load of teachers". Typical research methods use monitoring of the teacher's vocal load over several days and compare the results with in-field measurements of background noise, room acoustic parameters and self-reports of teachers and students followed by investigation on speech production accommodations due to changes in the acoustic environment. Other studies have shown the impact of classroom on health and social behaviour of teachers, in terms of discussions on stress level and work demands of teachers. Research in the area of occupational voice problems and voice ergonomics has aimed also on interplay between the individual and the work environment that emerge the voice problems. Different studies highlighted the confirmation of the relation between the poor indoor air quality and increased occurrence of laryngitis. This article reviews available recent information on voice and classroom interactions and discusses how vocal effort of teachers should be included in acoustical guidelines.

INVITED

Acoustics in Education:

Paper ICA2016-828

The impact of using different time sampling windows on measured noise metrics in K-12 classrooms

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Abstract

This paper will provide an overview of the current research at the University of Nebraska—Lincoln which aims to establish how indoor environmental conditions, including acoustic, indoor air quality, thermal and lighting conditions, in K-12 classrooms impact scholastic achievement. A largescale in-situ survey is being undertaken to gather data on indoor environmental conditions during multiple seasons (fall, winter, and spring). This paper will report noise metrics and reverberation times gathered to date from 110 classrooms. In particular, data logged over multiple school days are used to investigate the degree to which measured occupied and unoccupied noise metrics (such as equivalent A-weighted sound level, statistical levels such as L10 and L90, and occurrence rates) are affected by the length of time used for the metric calculation, or time sampling window. Recommendations for when and over what time period acoustic measurements should be made for evaluation over the course of regular school days in K-12 classrooms will be discussed.

INVITED

Education in Acoustics:

Paper ICA2016-724

A quasi-experimental field study of the impact of classroom noise on adolescents' mathematical performance

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Abstract

Two experiments investigating the effects of realistic levels of classroom noise on secondary school pupil's performance on mathematical, working memory and (non-verbal) speed of processing tasks are reported. Nine-hundred and seventy-six pupils, aged 11- to 16-years, completed bespoke tasks on laptop computers measuring numerical reasoning, arithmetic, working memory span and speed of information processing while classroom noise derived from classroom recordings was presented over headphones. In Experiment 1, classroom noise was presented in two conditions reflecting the maximum (70 dB) and minimum (50 dB) levels of classroom noise observed as secondary school pupils engaged in individual work. Experiment 2 assessed the effects of a more moderate 64 dB with the baseline condition of 50 dB. In Experiment 1, performance was statistically significantly poorer on the mathematical and working memory tasks in the 70 dB condition; however, there were no significant differences between conditions on the speed of processing task. In Experiment 2, there were no significant differences between performances in the 50 dB and 64 dB conditions. Results are discussed in terms of unfavourable classroom noise levels negatively impacting verbal processing of information and manipulation of information in working memory during problem solving.

INVITED

Education in Acoustics:

Paper ICA2016-83

Active learning in the "Analysis and Synthesis of Musical Signals" course

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Abstract

The electrical engineering bachelor course at the University of Campinas, Brazil, offers 13 areas of specialization, one of them being Sound Engineering. This specialization provides students with basic knowledge on engineering tools and applications related to sound and audio. To be eligible for this certificate the student has to participate in a total of 16 credit hours of courses related to classical acoustics, digital signal processing for audio, and musical theory (musical notation and perception). One of the core courses in this program is "Analysis and Synthesis of Musical Signals". At the end of this course, the student should be able to design a system for audio signal analysis and synthesis, and properly adjust the system's parameters for a particular purpose. In the first semester of 2016, this course was reorganized based on "active learning" and "project-based learning" methodologies. In this manuscript, we discuss the motivations for this change and describe how these active methodologies were employed. Also, we discuss how they contributed to the process of empowering students to learn mostly on their own while conducting project work.

Education in Acoustics:

Paper ICA2016-763

CINTRA: From an interdisciplinary centre on acoustics towards an intra-inter synergetic network

Claudia Arias[a;b;c], Aldo Ortiz Skarp[a;b], Mercedes Hüg[a;b;c], Jorge Pérez Villalobo[a], Sebastián Ferreyra[a], Fernando Bermejo[a;b;c], Laura Fernández[a;b], María Hinalaf[a;c;d], Fabián Tommasini[a;b], Valentín Lunati[a], Pablo Kogan[a;e], Agustín Cravero[a], Guillermo Gilberto[a], Marina Cortellini[a]

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Abstract

Acoustics is an inherently interdisciplinary science that has evolved in a mature discipline field occupied in the study and solution of hot social topics. This is, precisely, the strength and the challenge of the Acoustics in the XXI century. On the other side, nowadays it is out of discussion that universities have a crucial role in knowledge generation with social responsibility towards a more equalitarian society. In this direction, in Argentina new scientific policies promote three fundamental dimensions: a) interdisciplinary and inter-institutional articulations, b) an unified science and technology system, c) coupled with education in all levels and the productive sector. In turn, new paradigms of higher education gives to the research training abilities a central place in the academic curricula. In this complex scenario, a research centre has a key role in the achievement of those main aims. CINTRA, FRC, UTN - UA CONICET, is one of the seven Argentinean laboratories devoted to Acoustics that depends on public universities. In accordance with these lines of thoughts, it is dedicated to interdisciplinary research; young researchers training; education in acoustics and specialized technological service. At present, CINTRA is involved in a collective construction pointing to innovative dynamics encouraging the emergence of synergetic networks where theory and practice are naturally jointed. In this paper, we describe three ongoing programs that articulate research - technological innovation - interdisciplinary researcher training - technological services.

Wednesday morning, 7 September 2016 **Room 204**

09:00 - 10:40

Structural Acoustics and Vibration

SV2 - Structural Acoustics and Vibration (others)

Structural Acoustics and Vibration (others):

Paper ICA2016-182

Generation of virtual sliding resistance on a glassy panel using vibration

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Abstract

The problem of writing or drawing using the electronic pen as an input device on smart devices is the slip on the display panel due to the low sliding resistance between panel and stylus. The writing perception and the script style on the smart device are different from those using a pencil on the paper. This study is about a method for generating a virtual resistance on the glass panel to be identical with that using a pencil on the paper. It is thought that the dynamic friction can compensate the friction deficit by appending the normal vibration, in addition to the friction due to material characteristics of pen and glass. However, the question about the effective dynamic friction value of the oscillatory excitation, being thought as zero, remains. In this work, the answer to this query is investigated through the experiments. A prototype stylus having a stack of piezo actuators behind the tip is developed. Experiments are conducted and the measured data are used as the reference: the

normal force range in writing, the range of writing speeds, the friction coefficients at two sets of contact conditions, i.e., pencil-paper and plastic pen-tipglass panel, and the usual spectrum in writing scripts by a pencil on a paper. The effective frequency range is measured as 500-1200 Hz. For the stylus and glass set, the normal force amplitude that should be compensated for realizing the same writing feeling with the pencil and paper set is at least 0.11 N for the sliding velocity of 12-33 mm/s. The vibration signal having the peak envelope of 0.13 N is fed to the actuator in the stylus to generate the virtual resistance feeling. Subjective test with 10 subjects reveal that they can clearly perceive the increased sliding resistance.

Structural Acoustics and Vibration (others):

Paper ICA2016-99

Estimation of the vibration response of a plate with unknown boundary conditions using the iterative variation method

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Abstract

For controlling the vibration field of a structure by using a proper weighting on the control actuators, one needs a distribution of vibration response created by excitation forces injected from the actuators before or during the control actions. However, considering the time drift of physical properties and environment-related conditions of the controlled structure and actuators, such information would cause the error in control and might result a spill-over eventually. Therefore, when the control is to be performed in frequent machine on-off condition, we need a precise and quick method for estimating the vibration transfer functions at every time just before the control. To this end, we adopt the iterative variation method to quickly approximate the solution by utilizing some known responses by a preliminary measurement. The precision and speed in obtaining the solution depend heavily on the number of unknown coefficients and iterations. A thin rectangular panel with unclear boundary conditions, 400x900 mm^2 in size, and subject to a single point excitation is selected as the test example. Comparing the estimated and measured responses, in which 171 known responses are used as initial data, the error is found to be less than 6% for an average for 3600 points.

Structural Acoustics and Vibration (others):

Paper ICA2016-387

Study of the modal parameters of a wind turbine runner able to influence its acoustic behaviour

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Abstract

Wind turbines designed for small installations have to be able to harvest energy from low velocity winds, the most frequent in many areas. A natural solution to this challenge consists in choosing high solidity runner design, with increased number of blades. This may come together with a ducted runner design, eventually provided with a wind concentrator, able to collect wind from a larger area than that swept by the blades themselves. This approach was considered for the wind turbine studied in this paper. The quite complex design solution for the wind turbine meant also the use of different materials for every component, in order to best meet the specific structural requirements. The blades have steel tube spars and metallic or glass fibre reinforced composite skin and ribs, the outer cylinder/ring ducting the blades is made of sandwich material, while the radial bars sustaining the blades and the outer cylinder are made of steel again. This high solidity six blade ducted runner was subject of a comprehensive modal analysis, which put in evidence the natural vibration modes and associated frequencies, in various design variants. The results obtained will offer the possibility to assess the structural integrity, when damages

affecting various parts of the structure will consistently modify the modal behaviour, and to evaluate the propensity towards uncomfortable acoustic emissions, in various running conditions.

Structural Acoustics and Vibration (others):

Paper ICA2016-516

Guided convected acoustic wave coupled with a membrane wall used as noise reduction device

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Abstract

It is known since long that an essentially plane guided acoustic wave coupled with a vibrating wall could be attenuated in a certain frequency range. Various types of yielding walls can be studied such as plates, membranes or shells. This paper will focus on an acoustic wave convected by a uniform flow, propagating in a duct and interacting with a membrane in a 3 dimensions configuration. The problem is solved in the domain with the yielding structure via a home-made finite element model and coupled with a pure analytical plane wave beyond. It will be shown numerically that attenuation with flow reduces the attenuation efficiency. As results have been published for a similar (but not totally identical) configuration for small Mach number (numerically with other methods, and experimentally) the comparison is possible.

Structural Acoustics and Vibration (others):

Paper ICA2016-570

Material properties and microstructure contributions to vibrational damping in arundo donax L: Reed cane for woodwind instruments

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Abstract

Natural cane reeds (Latin name Arundo Donax L and here termed ADL) have been used on woodwind instruments for centuries with little change. The reed acts as a mechanical valve controlling the energy input into the musical instrument and it is the musician's first option for altering the instrument's sound and response characteristics. Despite this, their consistency, variable performance, durability and sensitivity to ambient conditions make it difficult for the musician to find and maintain a reed that responds to their liking. Manufacturers control the geometry of the reed to a high degree of repeatability, minimizing the influence of geometry on inter-reed variability. Thus it is desirable to examine the material, microstructural and anatomical properties of the reed and their contributions to vibrational performance. In the present work raw samples of ADL obtained from a manufacturer in pre-cut form are sectioned into longitudinal and transverse specimens for mechanical characterization in three primary directions. Measures of material damping (internal friction) are obtained through observations of phase lag between input stress and output strain waveforms. The effects of mechanically induced vibrations on these measurements are also investigated through the use of a shaker rig inducing small amplitude, fixed-free vibrations up to 1000Hz. Experimental samples are examined under optical microscope for characterization of several microstructural features. It is shown that the fiber size and specimen orientation contributes to the measured values of damping (tan δ). Mechanical vibrations simulating in-vivo conditions are also found to affect tan δ values with time. Future work will include the comparison of the results with those obtained from reeds that are regularly played by a professional musician.

Structural Acoustics and Vibration (others):

Paper ICA2016-618

Future tunnel railway underneath a city. Train-buildings transfer functions estimate and at home vibration and noise prediction

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Abstract

When designing a new railway line in a in a city, close or underneath buildings with human occupation, a study for the prediction of the vibrations induced by the trains traffic, and evaluation of the subsequent human discomfort is mandatory. This paper relates the soil and buildings vibration transmission estimates, done for buildings of mainly home apartments, in a city where a new extension of an underground railway line was designed and is being constructed, and the prediction of the vibrations caused by the future railway. The present work was based on the methodology proposed in previous papers [1, 2, 3] of the same authors, adapted [4] to the field and to the means available. Prediction of the vibrations near the railway was done by transposing the measurement values of the vibrations induced by underground trains going in an existing section of the line that is now being extended. Vibration transmission through the soil was estimated by measurements performed in the tunnel and on the ground surface in selected places, near the foundations, and on upper floors, of the selected buildings likely to undergo vibrations above acceptable limits [4] and possibility of resonances in the building floors was accounted for. Sensitive accelerometers and impulsive forces, these produced by excavation equipment in the tunnel, were used. Measurements were performed with the tunnel after excavation and in two points in the tunnel in a more advanced stage, with the isolation system and rails and slabs already in place. Attenuation was evaluated using the frequency analysed measurements. Conclusions and comments are drawn from the results of the experimental work described.

Structural Acoustics and Vibration (others):

Paper ICA2016-798

On the use of shunted piezo actuators for mitigation of distribution errors in resonator arrays

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Abstract

Earlier work has shown that an array of very small attached resonators can be designed to alter the dynamic response of a primary structure. The altered response can be designed to make the primary structure appear heavily damped or to have a particular spectral shape such as a bandpass response. However, small errors in the distribution of mass and stiffness distribution of the attachments can have a significant effect, degrading the intended performance. This presentation

discusses a concept of correcting small property distribution errors using shunted piezoelectric strip actuators bonded to the attachments.

Structural Acoustics and Vibration (others):

Paper ICA2016-824

On using the Hilbert transform for blind identification of systems with complex modes

Jose Antunes[(a)], Philippe Piteau[(b)], Xavier Delaune[(b)], Laurent Borsoi[(b)], Vincent Debut

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Abstract

The modal identification of dynamical systems under operational conditions, when subjected to wide-band unmeasured excitations, is today a viable alternative to more traditional modal identification approaches based on processing sets of measured FRFs or impulse responses. Among current techniques for performing operational modal identification, the so-called blind identification methods are the subject of considerable investigation. In particular, the SOBI (Second-Order Blind Identification) method was found to be quite efficient. SOBI was originally developed for systems with normal modes. To address systems with complex modes, various extension approaches have been proposed, in particular: (a) Using a first-order state-space formulation for the system dynamics; (b) Building complex analytic signals from the measured responses using the Hilbert transform. In this paper we further explore the latter option, which is conceptually interesting while preserving the model order and size. Focus is on applicability of the SOBI technique for extracting the modal responses from analytic signals built from a set of vibratory responses. Aspects of the theoretical formulation for complex SOBI using the Hilbert transform are clarified and a convenient computational procedure for obtaining the complex cross-correlation response matrix is developed. We show that the correlation matrix of the analytic responses can be computed through a straightforward Hilbert transform of the standard real correlation matrix typically obtained from measurements. Then, based on numerical simulations of a physical multi-modal system subjected to distribute random excitation, we assert the quality of the identified modal matrix and modal responses extracted using both the standard and the complex SOBI techniques. To perform such analysis, a simple and feasible physical device is proposed, which enables controlled levels of the modeshapes complexity, without introducing significant modal damping even for strongly complex modes.

Acoustics of Worship Spaces:
Paper ICA2016-124

The effect of noise criteria levels on speech intelligibility in various types of mihrab design

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Abstract
One of the most visible elements of mosque architecture that exhibits the quintessential of Islamic art and symbolism is the mihrab The shape and form of mihrab may be built using strong regional traditions or remarkable materials with decoration; to function as a directional guidance toward *Ka'aba* for Muslims performing prayers and where the *imam* (prayer leader) leads in a congregated prayer. In our previous research hypothesis, the simulated results for the four types of mihrab forms in six prototype models of mosques showed a little significance effects on their performance of speech intelligibility based on usage of the measured background noise during the previous measurement. Therefore the objective of this paper is to re-examine this hypothesis by selected degree of background noise levels in relation with their acoustical performance using computer modeling and acoustic simulation. Four noise criteria levels (NC) *i.e.* NC15, NC25, NC35 and NC45, were set into the prototype models of simulation configurations for examination on their speech transmission index (STI). The simulated results show the tendency in STI that the lower NC is, the higher speech intelligibility becomes. This paper presented a fundamental data to assist with future refinements.

Acoustics of Worship Spaces:
Paper ICA2016-213

Numerical and experimental acoustical analysis of a contemporary modern church

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Abstract
The acoustic project of a modern Italian church (Sant'Anna Church in Rapallo) has been set facing the typical concurrent needs of guaranteeing high speech intelligibility as well as a long reverberation time. Since the architectural design had already been finalized, the study has been developed by analysing different setups of internal finishes, in order to identify the configuration capable to obtain good acoustics. The values of several acoustical parameters were calculated by means of a geometrical acoustics based software for different design solutions, aiming to determine the best possible one. The modal analysis of the room was also carried out by means of a finite element model in order to verify that the eigenfrequencies to be evenly spread to the aim of obtaining a clear and neutral sound. This methodology was proven to be adequate and easy to replicate in similar design conditions.

Acoustics of Worship Spaces:
Paper ICA2016-269

Acoustics of Notre-Dame Cathedral de Paris

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Abstract
Notre-Dame de Paris is amongst the most well-known worship spaces in the world. Its large volume, in combination with a relatively bare stone construction and marble floor, leads to rather long reverberation times. Despite the notoriety of this space, there are few examples of published data on the acoustical parameters of this space, and these data are often not in agreement. Archived measurement recordings from 1987 were recovered and found to include several balloon bursts. In 2015, a measurement session was carried out which included similar source-receiver pairs using both balloon bursts and swept sine stimuli. Comparisons between results from these two sessions show a significant decrease in reverberation time in the modern state. This change is attributed to the addition of carpet in several areas of the cathedral. A geometrical acoustics model of the cathedral was constructed and calibrated from the 2015 measurements. The effect of carpeting was investigated through simulations. Comparison of the 2015 room impulse responses measured over the course of the 1 hour measurement session also indicated a potential slowly time-variant system. This was attributed to small temperature changes within the cathedral. Correction of this variance using a recently developed method allowed for the averaging of repeated measurements, providing the correct value for the reverberation time estimation, an improved signal-to-noise ratio, and a quantification of the temperature changes. This paper presents the results of these measurements, providing a modern documentation of the acoustical parameters of this historic worship space.

Acoustics of Worship Spaces:
Paper ICA2016-77

Acoustical redesign of “Nuestra Señora de La Paz”

Fernando del Solar Dorrego[a], Pablo Gardella[b]
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Abstract
Church "Nuestra Señora de la Paz" is located in Pilar, province of Buenos Aires. It has an octagonal floor plan and an internal volume of 2500 m3 with bare sidewalls and little ornamentation. Excessive reverberation (4 seconds at mid-band frequencies), echoes and an inappropriate sound system impaired speech intelligibility making ceremonies almost unintelligible. The church also suffered from thermal problems due to insufficient thermal insulation on the roof. Measurements (reverberation time, impulse responses and STI) were carried out and subsequent analysis revealed the significance of these problems. A holistic acoustical and thermal reform was proposed and a new sound system was designed to address these issues. Subsequent measurements validated the effectiveness of the proposed reforms.

Acoustics of Worship Spaces:
Paper ICA2016-591

Review of aspects that shape the aural experience in worship spaces

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Abstract
Physical measurements of architectural acoustics do not precisely reflect the human acoustical experience in worship spaces. While many studies focus on architectural acoustics, aural

architectural analysis that includes perceptual and cultural aspects, in addition to the physical aspects, and provides more comprehensive understanding of the aural experience, is afforded less attention. Worship spaces require complicated acoustical environments that allow both the hearing of sound clearly and the experiencing of sound aesthetics. This experience can create emotional effects through the perception of sound; therefore, there is the need for further study on the relationship between the qualitative and quantitative acoustical characteristics of religious buildings. This review paper provides a comparative analysis of previous studies in terms of physical, perceptual and cultural aspects of acoustics, and further clarifies the research gap in this area. Finally, it recommends strategies for studying the aural experience in worship spaces through the interaction between the cultural influence that affects how a specific sound is perceived, and the architectural elements that change acoustical parameters, thus playing an important role in the perception of the sound in the space.

Wednesday morning, 7 September 2016 **Auditorium 3**
Underwater Acoustics
09:00 - 10:40
UW1 - Underwater Acoustics

Underwater Acoustics:
Paper ICA2016-780

Arctic ambient noise: Synoptic measurements in the Fram Strait marginal ice zone

Dag Tollefsen[(a)], Hanne Sagen[(b)], Hans C. Tengesdal[(c)]
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Abstract
Measurements of underwater ambient noise in the Arctic are of recent interest due to changes in the ice cover and increasing human activity. A series of noise measurements were conducted in the Marginal Ice Zone (MIZ) of the Fram Strait (Greenland Sea) in the years 2010–12. This presentation gives an overview of experiments conducted with fields of sonobuoys deployed by P-3 aircraft over 150 km x 150 km in the MIZ from open ocean to compact ice under varying environment conditions and in different seasons. Noise spectra at frequencies from 10 Hz to 1 kHz are presented and compared with historical data. The spectra are categorized by ice zones and discussed with respect to environmental parameters that include sea state, wind force and direction, ice concentration, ocean swell, and sound propagation conditions. Anthropogenic and biologic components of the noise fields including sound due seismic exploration activity and marine mammals are also discussed.

Underwater Acoustics:
Paper ICA2016-917

Acoustic noise properties in the rapidly changing Arctic Ocean

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Abstract
The Arctic Ocean is undergoing dramatic changes, the most apparent being the rapidly reducing extent and thickness of the summer ice cover. As has been well established over prior decades, the environmental acoustics of the ice-covered Arctic is dominated by two major effects: the highly inhomogeneous ice cover, and the monotonically upward refracting sound speed profile, the combination of which forces all sound paths to be exposed to strong scattering loss and the associated loss of coherence. In some portions of the Arctic Ocean, however, a persistent inflow of a shallow 'tongue' of warm Pacific water has recently been discovered, which has also dramatically altered the acoustic environment, creating a strong acoustic duct between approximately 100 m and 200 m depth. This duct has the potential of trapping sound out to significant ranges (80 km - 100 km) without interacting with the ice cover, resulting in much higher coherence and signal preservation. Acoustic noise measurement results collected with a

vertically suspended array during ICEX 2016 illustrate the spatial and temporal noise properties in the presence of this acoustic duct at different depths. Comparisons of the ICEX 2016 data are also made with modeled Arctic noise data. [Work supported by ONR and DARPA].

Underwater Acoustics:

Paper ICA2016-427

Acoustical images of the Gulf of Gdansk

Eugeniusz Kozaczka[a], Grazyna Grelowska[b]

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Abstract

Acoustic images of seabed are of interest to specialists in the field of marine engineering, marine navigation, marine archeology, hydrogeology, etc. New technologies based on use of elastic waves, predominantly acoustic waves that allow detecting complexity of geometric forms of seabed, ensure progress in studying the bathymetric structure of seafloor. Use of parametric sources of waves generated by nonlinear interaction of collinear acoustic beams of little different frequencies is one of the most important factors allowing the study of the surface structure of a bottom. The paper presents the results of investigations performed in situ, for the same areas of interest, using multibeam sonar, side scan sonar and parametric sonar.

Underwater Acoustics:

Paper ICA2016-438

Seismic oceanography in Brazil from legacy MCS data of the oil and gas industry

Marcus Vinicius Carpes Barão[a], Berta Biescas Gorriz[b], Guillaume François Gilbert Barrault[a], Antonio Henrique da Fontoura Klein[a], Izabelle Aller[a], Marina Bousfield[a], João Paulo Ristow[c]

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Abstract

Seismic oceanography is a scientific discipline that studies the physics of the oceans using acoustic reflectivity data collected with multichannel seismic (MCS) systems. The seismic reflections generated within the boundaries of water masses with different thermohaline properties allow us to reconstruct images of the finestructure as well as mesoscale structures. This acoustic method can sample lateral sections along hundreds of kilometers to full ocean depth, with resolutions of 10 m -100 m in lateral and vertical dimensions, which covers an observational gap in physical oceanography. This work shows the detection of thermohaline finestructure, between 200 m and 1000 m deep, 100 km far away from the Santa Catarina Brazilian coast, using MCS data from the legacy data acquired in 1993, by the oil company Petrobras. The seismic data is compared with temperature and salinity data from the open access World Ocean Database and results support that the detected finestructure is generated by the mixing between the South Atlantic Central Water and the Antarctic Intermediate Water present in the area. Although the goal of the oil companies is to explore the solid Earth, the enormous legacy MCS data acquired around our planet and along decades contain significant information of the water column, which can help oceanographers to characterize the interaction between the mesoscale, submesoscale and fine-scale structures, as well as the interaction with the submerged topography and the energy cascading. Exploring these legacy data spread along decades might provide information about the evolution of temperature changes in the ocean and how climate change affects them.

Underwater Acoustics:

[Paper ICA2016-834]

Improving computation speed in shallow water reverberation modeling

Dajun Tang

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Abstract

Previously, a reverberation model has been developed for range-dependent shallow water environments [*J. Acoust. Soc. Am.* 131 (6), 4428- 4441 (2012)]. This physics-based model is formally exact under first order perturbation approximations. However, the computation speed is limited due primarily to two factors: (i) Fourier synthesis requires simulations at a large number of frequencies, especially when long range reverberation is concerned; and (ii) statistical averaged reverberation intensity requires a large number of interface realizations. Using bottom backscatter as an example, several techniques are presented which speed up the computation by simulating reverberation pressure field at a reduced number of frequencies and a reduced number of surface realizations.

Wednesday morning, 7 September 2016 **Auditorium 3**

11:00 - 12:00

Underwater Acoustics

UW1 - Underwater Acoustics

Underwater Acoustics:

[Paper ICA2016-636]

Guidelines for prediction and evaluation of acoustic impact on underwater fauna

Max Glisser[(a)(b)], Ismael Gómez[(b)(c)], Camilo Padilla[(b)], Patricio Priede[(b)]

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Abstract

For human beings, the most important sense is vision. In case of aquatic animals, hearing plays a fundamental role in the interaction with the surrounding medium. Unlike light, which dissipates quickly underwater, sound can travel farther and with greater speed under water than in air, allowing animals to communicate over long distances. Noise produced by human activities have the ability to modify the underwater acoustic environment, potentially altering natural processes such as mating, hunting and survival of living beings that inhabit it, even causing permanent hearing loss. The noise associated with construction activities of port sectors (such as underwater blasting, pile driving, welding, dredging, etc.) as well as seismic exploration, could have harmful effects on marine life. In this paper are reviewed the major technical and methodological issues involved in the assessment of noise impact on underwater animals, providing guidelines to address specific projects, considering, inter alia, acoustic descriptors, the differentiation between natural and anthropogenic underwater noise sources, sound attenuation factors in underwater environment, effects of noise on marine life, and commonly used control measures.

Underwater Acoustics:

[Paper ICA2016-138]

Comparison of tracking algorithms for dolphin whistle contour detection

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Abstract

The automated detection of marine mammal calls is a first and crucial step for many applications, such as real time monitoring, abundance estimation, mitigation and behavioral research. This work considers Gaussian Mixture Probability Hypothesis Density (GM-PHD) filter, an automated multi-target tracking technique, to track dolphin whistle contours. Over 9000 whistles from six different dolphin species are tracked. The performance of the GM-PHD detector is compared to the performance of the whistle detector from Pamguard, a widely used open source software tool. The performance of the detectors is similar in terms of precision, recall and mean deviation from the true whistle path. However, the GM-PHD detector has much better coverage of individual whistles and smaller fragmentation compared to Pamguard detector. Based on results, the GM-PHD detector appears to be a powerful tool for simultaneous tracking of odontocete whistle contours. It is computationally efficient, well suited for real-time implementation and applicable to different odontocete species.

Underwater Acoustics:

[Paper ICA2016-644]

Acoustic noise interferometry in a nonstationary ocean

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Abstract

Interferometry of underwater noise provides a way to estimate physical parameters of the water column and the seafloor without employing any controlled sound sources. In applications of acoustic noise interferometry to coastal oceans, the propagation environment changes appreciably during the averaging times that are necessary for the Green's functions to emerge from noise cross-correlations. Here, it is shown that the ocean nonstationarity limits from above the frequency range, where noise cross-correlations approximate the Green's functions. A theory is presented that quantifies the coherence loss of measured noise cross-correlations due to tidally induced changes in the water depth and gravity waves on the ocean surface. The theoretical predictions prove to be in a quantitative agreement with results of the 2012 noise interferometry experiment in the Straits of Florida.

POSTER

Numerical Techniques (others):

Paper ICA2016-45

Transmission loss optimization using genetic algorithm in Helmholtz resonator under space constraints

Gabriela Cristina Cândido da Silva[a], Maria Alzira de Araújo Nunes[b]
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Abstract

The development of high-performance mufflers associated to a compact volume it is of great importance in industrial field in order to obtain duct noise reduction in an economically and efficiently way. The Helmholtz resonator (HR), a classical reactive muffler, is considered in this work. The main purpose of this paper is to optimize numerically the sound transmission loss (TL) of the HR, since TL is an intrinsic characteristic of the muffler and does not depend on the source or termination impedances. The optimization methodology consists in maximizing the TL obtained for a pure tone frequency under dimensional constraints when the HR is considered mounted in a duct system. In order to search for the optimal dimensions of the HR cavity, an evolutionary search algorithm has been used. The sound pressure data is obtained from a finite element (FE) model (Ansys®) of the system HR/duct which communicates with the genetic algorithm (GA) implemented in Matlab® software. Many numeric simulations were performed varying GA parameters: number of generations and population size. The results show a strong match dependency between these parameters as well the importance of the bounds constraints range. Optimized HR has a gain of approximately 35 db attenuation in the frequency of interest when compared with the baseline.

POSTER

Numerical Techniques (others):

Paper ICA2016-668

Transient response prediction of an impulsively excited structure using a scaling approach

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Abstract

This paper is concerned with the transient response prediction of an impulsively excited structure. The frequency range typical for such problem is well beyond the practical limits of the standard finite element methods. In practice, the transient statistical energy analysis is often applied to the transient problem, but transient statistical energy analysis models and solvers are not readily available in some cases. Solving the problem in the framework of finite element analysis is still of practical value. The present work extends earlier scaling approaches for the steady state analysis to the transient response prediction. Based on the transient statistical energy analysis, a general scaling law is derived from the similitude of the governing equations of the scaled model and the original system. Because the scaled model is of reduced complexity and has similar dynamics to the original system, it can be efficiently solved with a transient finite element solver to simulate the impulsive response of the original system. Numerical examples involving coupled plate structures validate the applicability of the proposed approach. Due to its computational efficiency, the approach offers a way to extend the finite element methods to the impulsive response analysis.

Wednesday morning, 6 September 2016 Lounge Lateral Room
11:00 - 11:20
POSTER SESSION - Monitor 1
Architectural Acoustics - Room and Building Acoustics
AA2 - Acoustics of worship spaces

POSTER
Acoustics of Worship Spaces:
[Paper ICA2016-331]

Analysis of acoustic variations due to different use configurations in the "Basilica de Santa María de Elche", venue of the "Misteri"

Ana Planells, Jaume Segura, Arturo Barba, Salvador Cerdá, Rosa Cibrián, Alicia Giménez
Grupo de Investigación en Acústica Virtual Universitat Politècnica de València-Universitat de València. Edificio 5D, Camino de Vera s/n 46022. Valencia-Spain, acusvirt@upv.es

Abstract
Worship spaces are characterized by their use versatility. They are meeting places where the acoustics should be adequate for speech intelligibility but also suitable for music. Therefore, it is important to know the acoustic behavior of these spaces in both cases and the differences that occur depending on the activity being performed.In the case of the "Basilica de Santa Maria de Elche", variations in the use of space are particularly important because in addition to worship and concerts, it is the scene where the "Misteri d'Elx" is represented. This medieval play was declared a National Monument in 1931 and included in the first Proclamation of Masterpieces of the Oral and Intangible Heritage of Humanity by UNESCO in 2001.In this paper we have carried out an acoustic simulation of the different configurations of the temple related to the activities usually developed in it in order to determine how these settings affect its acoustic behavior.

Wednesday morning, 6 September 2016 Lounge Lateral Room
11:20 - 12:00
POSTER SESSION - Monitor 1
Education in Acoustics
ED1 - Education in Acoustics

POSTER
Education in Acoustics:
[Paper ICA2016-366]

Practical class on room acoustic for sound and recording degree program in Santa Fe, Argentina

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Abstract
In this article the final work of students of the course in Room Acoustics for Sound and Recording degree program of the Instituto Superior de Música from Universidad Nacional del Litoral, Argentina, is described and results are shown. They designed, implemented and aurally tested pseudo random sequence diffusers. One group of students designed a one-dimensional quadratic residue diffuser, the second group an MLS and the third group a two-dimensional quadratic residue diffuser. They tested aurally for themselves and then in a practice shared with all the class in the recording studio facilities. The students experienced how to build diffusors and how do they sound.

POSTER

Education in Acoustics:

[Paper ICA2016-477]

Elective Sound and Habitat in the architect´s career

Beatriz Garzón[(a)], Elisa Soldati[(b)], Leonardo Paterlini[(a)], Natalia Cerasuolo[(b)], Agustina Gayer[(b)].

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Abstract

This paper aims to show the experience carried out in the elective subject matter "Sound and Habitat" of the Environmental Conditioning II Chair belongs to the Environmental Conditioning Institute of the Architecture and Urbanism Faculty of the Tucuman National University, created in 2014. The strategy of the teaching-learning process developed is part of the participatory action-research methodology and it implements the constructivist model and the inductive method for the consideration of the acoustic concepts and aspects at architectural and urban scales, including theoretical knowledge and practical skills with the intervention into a concrete reality. Among the achieved results can be highlighted the works about Architectural Acoustics, Noise Pollution and Soundscape in San Miguel de Tucuman City and also the organization of the First "International Noise Awareness Day" in Tucumán.

Wednesday morning, 6 September 2016 — **Lounge Lateral Room**

10:00 - 10:20

POSTER SESSION - Monitor 2

Physical Acoustics

PA2 - Sonochemistry and Sonoprocessing

POSTER

Sonochemistry and Sonoprocessing:

[Paper ICA2016-922]

An ultrasonic approach to investigate the effect of aeration on rheological properties of soft biological materials with bubbles embedded

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Abstract

In this paper, we present the results of our ultrasonic experiments aimed at examining the effect of introducing air bubbles, during the mixing process, on the rheological properties of soft biological materials such as flour doughs. The rheological properties probed using low intensity ultrasonic technique operated in transmission mode. The sample investigated included dough made from all-purpose flour. During the mixing stage, the powder-like material (flour) is transformed into cohesive dough comprised of the continuous dough matrix and air pebbles. The rheological properties of such materials determine the quality of the end cereal product. Ultrasonic parameters such as velocity and attenuation coefficient are sensitive to properties such as the size of the occluded bubbles, and hence have great potential of providing quantitative evaluation of the properties of such materials. Large changes in the complex longitudinal modulus (and hence the rheology of the dough) are associated with changes in bubble volume fraction. Examination of the frequency dependence of ultrasonic velocity and attenuation in the dough showed that at higher frequencies, velocity differences between vacuum mixed dough and dough mixed at atmospheric pressure were small, reflecting sensitivity to dough matrix properties. A broad peak in attenuation at approximately 1.5 MHz, along with the associated changes in the phase velocity, has the potential to provide information on bubble sizes in dough, an important outcome in

probing the properties of this opaque material. The results of this work ultrasonic techniques provide a useful tool for studying the effects of bubbles in opaque food systems such as dough, and can provide unique insights into events taking place in food processing operations.

Wednesday morning, 6 September 2016 **Lounge Lateral Room**
10:20 - 10:40
POSTER SESSION - Monitor 2
Physical Acoustics
PA3 - Ultrasound

POSTER
Ultrasound:
Paper ICA2016-221

Hybrid levitation apparatus for laboratory X-ray diffraction study

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Abstract
Levitation or containerless processing techniques including acoustic, aerodynamic, electromagnetic, and electrostatic levitations techniques in the field of materials research and technolgy are useful for the study of undercooling, nucleation, vitrification, phase equilibria, metastable state formation, microstructure formation, and synthesis of new functional materials and so on. We present a hybrid levitation apparatus based on both acoustic and aerodynamic levitation techniques designed for X-ray diffraction experiments using laboratory X-ray radiation source. The aerodynamic levitator employed a conical nozzle with a range of 0.6 ~ 1.0 mm in diameter aperture using aerodynamic forces to levitate a spherical type of solid specimens. The acoustic levitator which is composed of a single-axis transducer for generating ultrasonic sound with a frequency of around 20.3 kHz and a concave reflector assists to levitate the specimens in the vertical direction. Finally the hybrid (acoustic/aerodynamic) levitator was integrated with a laboratory X-ray diffractometer. In order to demonstrate whether this hybrid levitation technique can apply for X-ray diffraction study and collect a valuable diffraction data using the laboratory X-ray source, we carried out X-ray diffraction experimental for ball-shaped solid ZrO_2, Al_2O_3 and Si samples, showing the apparent density ranged from 2.3 to 5.7 g/cm3.

Wednesday morning, 6 September 2016 **Lounge Lateral Room**
11:00 - 12:00
POSTER SESSION - Monitor 2
Physical Acoustics
PA3 - Ultrasound

POSTER
Ultrasound:
Paper ICA2016-344

Acoustic emission sensors for GFRP wind turbine blade

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Abstract
This paper describes the fabrication piezo-electric acoustic emission(AE) sensor for glass fiber reinforced plastic(GFRP) wind turbine blade. The AE sensors mainly consist of piezo-electric element, front matching layer, and back acoustic material. As an active element for detecting AE signals from

GFRP, PZT, 1-3 composite was selected because its acoustic impedance was relatively lower than PZT ceramics. For the front matching material of AE sensor between surface of GFRP wind turbine blade and piezo element, the plexiglass was used. Considering the wide band frequency characteristics of AE sensor, high acoustic impedance backing material was fabricated. In order to evaluate the performance of developed PZT based 1-3 composite AE sensor, the pencil lead breaking test (ASNT standard E1106-86) was performed and its perform and its performance was compared with commercial PZT AE sensor. From the experimental results, the PZT based 1-3 composite AE sensor showed better performance than commercial PZT AE sensor.

POSTER

Ultrasound:

Paper ICA2016-348

A complete medical ultrasound imaging simulation based on COMSOL® and MATLAB®

R. J. Simões[(a)], A. Pedrosa[(a)], W. C. A. Pereira[(b)], A. V. Alvarenga[(c)], C. A. Teixeira[(a)]

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[(c)] INMETRO, Brazil

Abstract

Ultrasound (US) imaging is widely used in medical diagnosis. The development of tools to foster new high-quality ultrasound imaging is essential to improve medical diagnosis. A straightforward and low-cost way to speedup development is by exploring accurate computational simulations. Here, we propose a COMSOL R and Matlab R simulation that implements the stages involved in the formation of conventional ultrasound images. The simulation was developed using realistic medium properties, improving reliability, and aiming to establish a base for realistic, and further elaborated simulations, leading to a reliable evaluation of different US imaging parameterization and wave propagation. We simulate a transducer with 256 piezoelements, backing and matching layers that are excited by a four-cycle pulse of 1-MHZ central frequency. The resulting RF data is then exported to Matlab and a beamforming is made, resulting in a final reliable US image. We simulate ultrasound reflection from a geometry similar to the 100- millimeter AIUM test object - rod group E. The results suggest similarities with literature, and metrics like signal-to-noise ratios improve from -8.261dB to -3.809dB while entropy has a change from 5.758 to 3.385. These results trend in same way as in real scenarios.

POSTER

Ultrasound:

Paper ICA2016-819

On the possibility of non-invasive tissue assessment using induced changes in backscattered energy: A k-wave simulation

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Abstract

Ultrasound (US) propagation properties change with temperature. One US feature that was studied is the change in backscattered energy (CBE). Theoretical models and experiments have shown that different types of tissues, made of different scatterer types, when submitted to a temperature change can produce distinct CBE. Thus, one could raise the hypothesis that CBE could be also useful for non-invasive tissue assessment. In this study a toolbox, k-Wave (www.kwave. org), is used to simulate CBE for different tissue types and properties. The implemented geometry consists of a 2D geometry simulating a water medium. Two distinct highly scattering regions were designed within this aqueous medium: one region of lipid-based scatterers (simulating fat) and one region of water-based scatterers

(simulating muscle). Backscattered signal profiles for temperatures within the range 37 ºC to 41 ºC were computed. The simulated US transducer was excited at 7.5 MHz. Amplitude variations of the RF signals were evaluated for the different temperatures by a linear fitting. As expected and shown by previous studies fat and muscle scatterers regions have shown a positive and negative variation, respectively. Our results show a positive variation of 0.75 dB·ºC^{-1} and a negative variation of –0.11 dB ºC^{-1}, as seen in literature for fat with density of 950 kgm^{-3} and muscle with density of 1065 kg·m^{-3}, respectively. Changing muscle density to 1085 kg·m^{-3}, the CBE variation was –0:09 dB ºC^{-1}, and changing fat density to 920 kg·m^{-3}, the CBE value increases in a rate of 0.75 dB ºC^{-1}.The developed simulation reinforces that idea that CBE could be valuable for non-invasive tissue classification. Future studies will include the validation of the simulation with data collected from laboratory experiments.

Wednesday morning, 7 September 2016 **Lounge Lateral Room**
09:40 - 10:00
POSTER SESSION - Monitor 3
Physical Acoustics
PA3 - Ultrasound

POSTER
Ultrasound:
Paper ICA2016-521

Tuning to a particular acoustic whispering-gallery mode in the GHz range

Sylvain Mezil[(a)], Kentaro Fujita[(b)], Motonobu Tomoda[(b)], Matt Clark[(c)], Oliver B. Wright[(b)], Osamu Matsuda[(b)]
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Abstract
Surface Acoustic Waves (SAWs) are commonly used in non-destructive testing and GHz filtering. Typical setups to study SAWs in the GHz range in the time domain make use of sub-picosecond light pulses. The absorption of pump light by the medium generates surface acoustic waves. The latter are detected by delayed probe light pulses through strain-induced variations in the optical phase. The spatiotemporal evolution of the SAWs is accessible by scanning the focusing position and time delay of the probe light pulses. In general, the laser repetition rate f_{rep} (typically ~80 MHz) limits such a setup to measurement frequencies that are integral multiples of f_{rep}. Commonly used setups focus pump light to a circular spot of a few microns in size, thus generating SAWs propagating in all directions. In the case of whispering-gallery modes (WGM) on a disk, that is, modes propagating around the disk rim, two counter-propagating modes are thereby excited with the same intensity. Here we overcome these two limitations. To access any arbitrary frequency, we modulate in intensity both the pump and the probe beams and then carry out appropriate analysis on lock-in detected probe-beam intensity variations. This opens the way to determine the acoustic dispersion curve of samples with arbitrary resonance frequencies as well as the quality factor of any chosen mode. In order to select only WGMs propagating in one direction, we make use of a spatial light modulator (SLM) programmed with the use of computer-generated holograms. In the particular case of WGMs, a windmill-shaped surface source pattern is chosen to produce acoustic WGMs with one rotation sense or the other. These new results extend the possibilities of SAW imaging by allowing fine control of excited surface acoustic modes.

POSTER

Speech Communication:

Paper ICA2016-786

Detailing speech processing in infancy using cortical auditory evoked potentials and multidimensional scaling

Kathleen McCarthy[(a)], Katrin Skoruppa[(b)], Paul Iverson[(c)]
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Abstract

Studies on infant speech perception have typically focused on individual phonetic contrasts (e.g., /i/-/y/), because collecting data on even a single contrast is very time intensive. The present study used an efficient measure of perceptual sensitivity to map perception across the British English vowel space for 80 monolingual English infants (4-5, 8-9 and 10-11 months old). Cortical auditory evoked potentials were measured for spectral changes between concatenated vowels, which, for infants, typically evokes a positivity about 150-200 ms after each spectral change. These were measured for 21 pairs of seven monophthongal vowels (/i/, /ɪ/, /ɛ/, /a/, /ɑ/, /ɔ/, /u/) that were presented in a random concatenated sequence with changes every 300-400 ms. ERPs were averaged across epochs following each spectral change, with the magnitude of the response for each vowel pair used as similarity measure for multidimensional scaling. The 4-5 month old infants had two-dimensional perceptual maps that closely matched the F1 and F2 acoustic differences between vowels. In contrast, the older infant response was less related to the vowel acoustics. They had selectively larger responses for spectrally similar vowel pairs (e.g., /i/-/ɪ/), suggesting a shift to a more phonologically driven processing. These results provide a more detailed picture of phonetic development than has been shown before, and demonstrate an efficient technique that can successfully map speech processing in infancy.

POSTER

Speech Communication:

Paper ICA2016-185

Effect of consonant manner on L2 speech perception in reverberation

Hinako Masuda
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Abstract

Perceiving speech in real life requires the ability to detect relevant acoustic cues among various linguistic or non-linguistic disturbance such as background noise and reverberation. Accurate perception in such environment can be challenging for even native listeners, and the challenge becomes even bigger for non-native listeners. This study investigated the perception of 23 English consonants in reverberant or reverberant and noisy conditions by Japanese learners of English and the effect of consonant manner on accurate identification. Participants were presented with twenty-three American English consonants /b ʧ d f g h ʤ ʒ k l m n p ɹ s ʃ t θ ð v w j z/ embedded in the context “You are about to hear a__a”. All participants listened to the stimuli in the order of 1) reverberant environments of RT = 0.78 s (D50 value 67.5%), 1.12 s (D50 value 47.7%), and 1.43 s (D50 value 32.2%) in randomized order), 2) noisy and reverberant condition (SNR = 10 dB added to reverberation RT = 0.78 s), and 3) the quiet condition. Participants listened to each stimulus and were asked to choose one of the consonants out of the 23 choices that was most similar to what they heard. Accurate identification rates of the Japanese listeners are compared to that of the English native listeners. Analysis was carried out as Language (Japanese or English), Manner (stops, fricatives,

affricates), and Condition as independent variables. Statistical analysis demonstrated significant interactions between language and condition, as well as language and manner.

Wednesday morning, 6 September 2016 **Lounge Lateral Room**
11:00 - 12:00
POSTER SESSION - Monitor 3
Speech Communication
SC1 - Speech Communication

POSTER
Speech Communication:
Paper ICA2016-738

A 3D computer game for testing perception of acoustic detail in speech

Daniel Duran[(a)], Natalie Lewandowski[(b)], Antje Schweitzer[(c)]
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Abstract
We present a novel experimental framework for perception studies and an application focusing on attention to fine phonetic detail in natural speech perception. Traditional psychological experiments in research on speech perception do not provide a natural testing scenario (notorious supervision and lack of naturalness). A solution to this problem is employing a computer game in which attention to fine phonetic detail comes natural. Computer games are increasingly used in psychology or in studying emotional speech production, where the communication in multiplayer games is recorded. Our novel framework implements a traditional psycholinguistic AB test paradigm within a computer game. Using a state-of-the-art game engine, we developed a first person shooter. This genre is ideally suited to implement a test scenario which requires the subjects to click on a specific point on the screen as fast as possible. The player moves around within a virtual 3D environment and reacts to stimuli presented by enemies which belong to two different categories, each of which is associated with one response key. The two categories are initially distinguished by visual and acoustic cues (e.g. different colors, and different sounds). Gradually, visual cues are removed. Thus, the subject has to attend to the acoustic cues and react accordingly. An additional important aspect of our framework is the high involvement in the game and motivation of the subjects to solve the task. In traditional psychological experiments, on the other hand, subjects may easily get tired or bored by the repetitive, unnatural task. We discuss practical and theoretical challenges encountered with the implementation of a psychological test within a computer game.

POSTER
Speech Communication:
Paper ICA2016-300

Voice Conversion of emotional speech using hidden Markov model-based speech recognition and synthesis

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Abstract
This paper describes a many-to-one voice conversion (VC) technique that does not require a parallel training set of the source and target speakers. A VC method consisting of decoding and synthesis parts that do not require a parallel training set has already been proposed. The basic idea of this system is that an input utterance is recognized utilizing the hidden Markov model (HMM) for speech recognition and the recognized phoneme sequences are used as labels for speech synthesis. Performance of this type of system depends on phoneme recognition accuracy. The aim of this work is to improve the performance of this type of VC system using a highly accurate acoustic model for

phoneme recognition. In particular, we focus on VC of emotional speech. In order to achieve this aim, emotional speech uttered by any speaker needs to be recognized. Speech recognition of emotional speech is a challenging task due to vast changes of acoustic features compared with normal speech. In order to solve the problem, we propose utilizing a deep neural network (DNN) as an acoustic model on the assumption that it is able to recognize emotional speech at a high level of accuracy. In order to evaluate the proposed system, subjective speech intelligibility tests were conducted on 8 subjects. The intelligibility was measured at the phoneme level. For the strongest emotion condition, the intelligibility scores were 78.7% and 72.9% with DNN-HMM and Gaussian mixture model (GMM)-HMM, respectively. The experimental results showed that the use of DNN-HMM contributed to the improvement of intelligibility scores in both normal and strong emotion conditions.

POSTER

Speech Communication:

Paper ICA2016-526

Investigation of the effect of articulatory-based second language production learning on speech perception

Atsuo Suemitsu[(a)], Takayuki Ito[(b)], Jianwu Dang[(c)], Mark Tiede[(d)]
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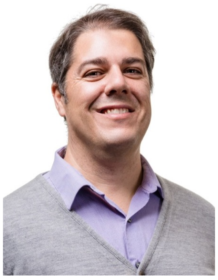
Abstract

The effect of second language production training on perception has been previously explored, but it remains unclear whether such training by itself influences the perception of speech sounds. In previous work participants heard the correct pronunciation of the target while simultaneously undergoing production training, making it unclear what component of improvement was due to the production training alone. In the current study we have therefore modified our electromagnetic articulometer-based training system, which provides estimates of learner-specific head-corrected tongue positions for a target utterance in real time, to eliminate simultaneous presentation of audio stimuli. Japanese learners of the American English vowel /æ/ performed ABX perceptual testing on this vowel before and after the visually presented articulatory-based pronunciation training. We examined whether or not the production-driven pronunciation improvement also induces a change in the perception of the second language sounds.

Wednesday midday, 7 September 2016 **Juan Pablo II Auditorium**
12:00 - 13:00
Plenary Lecture:
Chair: Alberto Behar



Frank Russo

Paper ICA2016-522

Understanding music perception from the perspective of oscillation and resonance

Frank A. Russo
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Abstract
Over the last decade my lab has investigated psychoacoustic properties of pitch, timbre, and rhythm as perceived by the ear (auditory) as well as the skin (vibrotactile). Mechanoreceptors in the skin are structurally similar to those in the ear and exhibit frequency tuning enabling coarse pitch perception. Although the skin is equipped with only a few broadly tuned frequency channels and without a “place code”, this appears to be enough to enable discrimination between complex vibrotactile waveforms that have been matched for fundamental frequency and subjective magnitude (i.e., vibrotactile timbre perception). The skin is also quite capable of giving rise to the perception of rhythm, however this capacity proves challenging with complex rhythms. Neuro-electric measures allow us to examine resonance to different levels of oscillatory structure. Auditory neurons in the brainstem are capable of phase locking with tone frequencies in music. The fidelity of this type of neural resonance is better in individuals with music training and worse in individuals with hearing impairment. Neurons in auditory and motor cortices have been found to phase lock to the dominant beat frequency in music (i.e., the pulse). This form of neural resonance continues even after the music has stopped, and much like the brainstem response to tone frequencies, its fidelity tends to be better in individuals with music training. The picture that emerges from this body of work is that perception of music is underpinned by neural resonance to different levels of oscillatory structure present in auditory and vibrotactile waveforms. Long-term active engagement with music supports the fidelity of neural resonance.

Wednesday midday, 6 September 2016 **Cardenal Pironio Auditorium**
13:00 - 14:30
Symposium GINER-CDM
Insulation and acoustic conditioning project of all the technical rooms of the new Disney Latin America headquarters in Argentina

INVITED

Wind Farm Noise:

Paper ICA2016-316

Wind turbine noise as experienced by impacted residents

Sveinulf Vagene[(a)], Willi Larsen[(b)]

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Abstract

In order to evaluate and mitigate the impact of wind turbine noise on residents living near windfarms, it is necessary to characterize and understand the noise-problem from the perspectives of both the scientist and the impacted residents. A comprehensive dataset consisting of more than two years of noise and wind measurements has been collected at a windfarm in Norway. A detailed noise diary was also recorded along with the sound data. We present an analysis where we relate the sound measurements to the dwellers noise diary. We also discuss the results in context of the pre-development noise mapping - and a study of health impact from the noise performed on the neighbouring residents of the windfarm. Our study shows a variable, and difficult to predict, sound propagation- and impact-pattern from the wind turbines. We demonstrate how the wind-shadow effect becomes a considerable factor in enhancing noise at receiver locations in mountain terrains. We conclude that conventional noise modelling of refracted sound from windfarms in such terrains is inaccurate, and should be discouraged as a tool for determining safe offset distances to windfarms. In such cases it will be safer to set a generous fixed distance between habitations and windfarms, rather than trust uncertain noise-modelling.

Wind Farm Noise:

Paper ICA2016-440

Physiological effects of wind turbine noise on sleep

Michael G. Smith[(a)], Mikael Ögren[(b)], Pontus Thorsson[(c)], Eja Pedersen[(d)], Kerstin Persson Waye[(e)]

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Abstract

In accordance with the EU energy policy, wind turbines are becoming increasingly widespread throughout Europe, and this trend is expected to continue globally. More people will consequently live close to wind turbines in the future, and hence may be exposed to wind farm noise. Of particular concern is the potential for nocturnal noise to contribute towards sleep disturbance of nearby residents. To examine the issue, we are implementing a project titled Wind Turbine Noise Effects on Sleep (WiTNES). In a pilot study described in this paper, we performed an initial investigation into the particular acoustical characteristics of wind turbine noise that might have the potential to disturb sleep. Six young, healthy individuals spent 5 nights in our sound exposure laboratory. During the final 3 nights of the study, the participants were exposed to wind turbine noise, which was synthesised based on analysis of field measurements. Exposures involved periods of different amplitude modulation strengths, the presence or absence of beats, different blade rotational periods, and outdoor $L_{AEq,8h}$=45 or 50 dB with indoor levels based on the windows being fully closed or slightly open. Physiological measurements indicate that nights with low frequency band amplitude modulation and $L_{AEq,8h}$=45 dB, slightly open window ($L_{AEq,8h}$=33 dB indoors) impacted sleep the most. The presence of beats and strong amplitude modulation contributed to sleep disturbance, reflected by more electrophysiological awakenings, increased light sleep and wakefulness, and reduced REM and deep sleep. The impact on sleep by these acoustic characteristics is currently the focus of interest in ongoing studies.

Wind Farm Noise:

[Paper ICA2016-444]

Characterization of the sound spectrum of the windregarding environmental studies, focused in wind energy devices

Pablo Gianoli[(a)], José Cataldo[(a)], Alice Elizabeth González[(a)], Joaquín Montero[(a)]

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Abstract

Wind energy is not only growing in Uruguay, but it is expected to continue increasing for many years. Environmental licenses are required for the installation of wind farms. In this frame, an accurate quantitative comparison of environmental conditions with and without power generation is needed a feedback for adjusting the design of measurement campaigns. Sound pressure levels were obtained in frequency bands of third-octave width under different conditions, in either places without wind turbines (data for the baseline characterization) and with wind turbines in different In this paper, the spectra of sound pressure levels of winds of different speeds are presented. They have been obtained by direct measurement both in wind tunnel and field. Both kinds of measurements allowed having operating situations. These measurements were complemented by other data performed in the wind tunnel of the Faculty of Engineering. It is a wind tunnel designed to simulate the atmospheric boundary layer, taking into account the characteristics of turbulence intensity, scales of turbulence and power spectra. Wind speed fluctuations need to be measured; the instrument used in wind tunnels to achieve these data is a hot wire anemometer at constant temperature. The most frequent winds in Uruguay are those from SW and NE. For the same wind speed, higher sound pressure levels occur when wind blows from the SW. Both the temporal evolution of sound pressure levels in scale A and in BTO respond to changes in wind speed. The main energy is found between 12,5 Hz and 630 Hz, with a 'floor value' of about 10 dBZ per band. For frequencies exceeding 630 Hz, negligible noise levels were found both in field and in laboratory. Thus, wind effects should be expected to occur for frequencies up to 630 Hz.

INVITED

Wind Farm Noise:

[Paper ICA2016-565]

Wind turbine noise source characteristics measured with a large microphone array

Stuart Bradley[(a)], Torben Mikkelsen[(b)], Sabine von Hünerbein[(c)], Mathew Legg[(d)]

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Abstract

A large 40 m scale microphone array was designed to record the noise from a wind turbine. The objective was to acoustically image the noise source characteristics across the entire diameter of the turbine at a spatial resolution of 1 m at 1/3 octave resolution. This allows simultaneous definition of the spatial, temporal, and spectral properties of the generated sound. The array comprised 42 purpose-designed low-noise microphones simultaneously sampled at 20 kHz. Very high quality, fast, meteorological profile data was available from nearby 80 m masts and from the turbine nacelle, giving wind speed, wind direction, and turbulence data. A speaker was mounted at the base of the turbine tower, for determining the spatial characteristics of coherence, and for compensating for local wind variations. An experiment was also run recording the sound from a continuous tone speaker mounted near the tip of a turbine blade, allowing testing of signal processing to correct for the very substantial Doppler shift. We describe the significant challenges in imaging with such a large array. High resolution image results are given as well as time-resolved and spectrally-resolved turbine noise directivity patterns.

Wind Farm Noise:

[Paper ICA2016-503]

Underwater noise mitigation from pile driving using a tuneable resonator system

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[(d)] Applied Research Laboratories & Department of Mechanical Engineering, The University of Texas at Austin, USA, pswilson@mail.utexas.edu

Abstract

This paper covers the development of a tunable acoustic resonance-based underwater noise abatement system for use with marine pile driving, offshore energy production, seismic sources, and ships, among others. The system consists of arrays of underwater air-filled resonators, which surround the noise source and are tuned to optimally attenuate noise in a frequency band of interest. Based on the predictive models for the acoustic dispersion relation in bubbly liquids given by Church and Commander and Prosperetti, this system has been shown to attenuate sound by up to 50 dB near the resonance frequency of the resonators. In this investigation, modeling and laboratory tests were used to tune the system for pile driving applications with a spectral noise peak near 100 Hz. System demonstrations that were conducted at two offshore wind farm construction sites in the North Sea will be discussed. In these tests, peak sound pressure level reduction of nearly 40 dB was measured around the design frequency, and almost 20 dB sound exposure level reduction was measured in the 20 Hz to 20 kHz band. The method of deploying these resonator arrays in a simple collapsible framework, as well as the operational advantages of this system for pile driving, will be described and details on a fully constructed noise abatement system that will soon be tested will be shared.

Wednesday afternoon, 7 September 2016 **Dr. Valsecchi Auditorium**
14:30 - 16:10
Signal Processing in Acoustics
SP1 - Acoustic Array Systems

Acoustic Array Systems:

[Paper ICA2016-598]

Sweet-spot-independent binaural reproduction with a listener-adaptive loudspeaker array

Marcos F. Simón Gálvez[(a)], Filippo Maria Fazi[(a)]

[(a)] Institute of Sound and Vibration Research, University of Southampton, Southampton, Hampshire, SO17 1BJ, United Kingdom

Abstract

Adaptive cross-talk cancellation systems require a smooth and fast update of the control filters for different listening positions, which is often achieved by simultaneous filtering and cross-fading. An alternative method is presented here, in which the control filters are calculated analytically in real-time, assuming each acoustical source is represented by a point-monopole radiator. This allows for a time-domain formulation of each filter as a sum of gain-delay elements, rather than as a single finite impulse response (FIR) filter. This leads to a very simple signal processing scheme, wherein the gain-delay elements are varied to adapt the response towards the listener. The performance of the proposed system is evaluated by means of numerical simulations with a geometry based on a 28 point-source loudspeaker array.

Acoustic Array Systems:

Paper ICA2016-686

Advanced delay-and-sum beamformer with deep neural network

Mitsunori Mizumachi[(a)] Maya Origuchi[(a)]
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Abstract

Signal enhancement can be achieved by spatial filtering with multiple microphones, that is, a microphone array. A traditional delay-and-sum beamformer enhances the target signal without spectral distortion, but could not control its beam-pattern, except for the arrival direction of the target signal. The beam-patten is flexibly designed by introducing channel-dependent weights or filters in delay-and-sum beamforming. The channel weights can be optimized using a neural network in order to achieve superdirectivity. In this study, an advanced delay-and-sum beamformer is proposed to make the main lobe much sharper and decrease grating lobes. The advanced beamformer substitutes a deep neural network for the conventional three-layered neural network. The proposed method also adopts a non-equally-spaced microphone arrangement and sub-band beamforming to prevent spatial aliasing. It is confirmed that the proposed beamformer has significant advantages both in sharpening the main-lobe and getting rid of grating lobes over conventional beamformers.

Acoustic Array Systems:

Paper ICA2016-415

Sound source separation in complex environments using an array-of-arrays microphone system

Jorge Trevino[(a)], Cesar Salvador[(a)], Virgilijus Braciulis[(a)], Shuichi Sakamoto[(a)], Yôiti Suzuki[(a)], Kyoji Yoshikawa[(b)], Takashi Yamasaki[(b)] and Kenichi Kidokoro[(b)]
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Abstract

Recording a specific sound source in a complex environment is challenging, especially when interfering sources lie between the target and the recording system. This can be avoided by placing a microphone next to the target so as to get a clear measurement. However, this may interfere with the events being recorded and lacks flexibility to select and track the desired source. Microphone arrays are used as adaptable systems to separate specific sounds given the spatial position of their sources. Arrays can be classified according to their geometry, such as linear or spherical arrays. In particular, spherical arrays are advantageous due to their compact shape. The highly symmetric design can also yield an almost constant angular resolution making their performance predictable. Unfortunately, this symmetry is also the source of their main limitation: sounds are measured from a single viewpoint. Spherical arrays can separate sounds according to their direction of arrival using a technique known as beamforming; however, it is difficult to separate two sounds when their sources and the array are aligned. The present research introduces a new approach to source separation using an array-of-arrays microphone system. The proposal, a cooperative variant of beamforming, relies on two or more spherical microphone arrays working together as a single, unified system. This yields a multiple-viewpoint measurement of the sound field and allows for sound source separation even when interfering sources are present between the target and each of the arrays. Such separation would be difficult using conventional beamforming. The proposal is evaluated through numerical experiments using both, physical models and real-world measurements for a system composed of two 64-channel microphone arrays.

Acoustic Array Systems:

Paper ICA2016-903

The optimization design of microphone array layout for wideband noise sources

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Abstract

Microphone array system has been an important tool to identify main noise sources from mixed acoustic field emitted from running machines. The performance of localization is affected by several parameters such as the microphone array layout, the number of microphone, weights and the array aperture. Usually the array aperture and the number of microphone are restricted in practical applications, and therefore array layout design and weights are the most crucial parameters to improve localization performance characterized by the array beam pattern. Array layout optimization design is highly related to frequency bandwidth. Therefore, a new array layout optimization method is presented in this paper for wideband noise sources to generate low side-lobe level beam patterns based on particle swarm optimization (PSO) which has proven to be very efficient in optimal layout design. In this paper, we present a general framework to consider the overall performance based on the narrow main-lobe width and low side-lobe level criterion using the particle swarm optimization method. In low frequency band we place more emphasis on the main-lobe width, and place more emphasis on the side-lobe level within high frequency band. In the presented paper, both weights and layout can be optimized simultaneously. Finally, simulations and experiments are carried out to demonstrate that the proposed scheme can improved localization performance of wideband noise sources.

Acoustic Array Systems:

Paper ICA2016-506

Impedance estimation of a finite absorber based on spherical array measurements

Antoine Richard[a], Efren Fernandez-Grande[b], Jonas Brunskog[c], Cheol-HoJeong[d]

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Abstract

A method to characterize the surface impedance of materials is presented. The estimation is based on pressure measurements with a spherical microphone array. These measurements are used to reconstruct the sound pressure and particle velocity on the sample's surface, from which the material's impedance is inferred. The accuracy of the reconstruction is improved by using compressive sensing, where the wave field is represented with only a few components, ideally an incident and a reflected wave. However, at low frequencies, diffraction from the edges contributes considerably to the soundfield. This leads to a deterioration of the impedance estimation, which is clearly visible in initial experimental results. The proposed methodology makes it possible to characterize the edge effect, and subsequently compensate for it in the processing, emulating measurements on an infinite sample.

Acoustics in Education:
Paper ICA2016-830

Good practices in music room acoustics in Brazil: A dissemination action and evaluation of its impact

Aloísio L. Schmid[a], Guilherme B.R.Romanell [b] Dinara X. Paixão[c], Gustavo S. V. Melo[d], Erasmo F. Vergara[e], Andrey R. Silva[f], Newton S. Soeiro[g], Letícia S. Rocha[h], Raquel R. Rocha[i], André Santana[j], Márcio H.S. Carboni[k]
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Abstract
The paper has the purpose of measuring the impact of an already finished research work aimed at establishing desirable acoustic conditions for music rooms and practice and rehearsal rooms in basic education schools in Brazil. The Brazilian Government determined that curricula of basic education, starting in 2012, once again include music as a mandatory subject. Therefore, there was the need to build new music rooms and practice rooms, and to refurbish existing ones. Abramus Project, which resulted from the cooperation of researchers in four Brazilian universities, had the purpose of exploring the opinion of acousticians and music teachers regarding the proper conditions for music rooms and practice (rehearsal) rooms, and of disseminating the results. The project's starting point was the lack of knowledge of school managers, as well as music teachers, regarding the desirable acoustic conditions of such rooms, including topics as gain, reverberation, diffusion, and insulation. Basic knowledge on Concert Hall Acoustics and Classroom Acoustics was explained; several schools were analyzed as case studies; absorption coefficients of furniture and objects found in classrooms were measured, and simulation with auralization was also applied. As a result, a richly illustrated book was issued and distributed to researchers, music teachers, school directors and public education managers. Finally, a survey was conducted on a sample of music teachers, regarding their experience with school architecture. Questions range from topics like "what do you know about room acoustics?" and "did you know the Abramus publication?" Results of the survey are going to be processed and presented. Authors expect new guidelines on how disseminate the academic knowledge on the subject.

Acoustics in Education:
Paper ICA2016-585

Acoustic evaluation of the design process and construction of the television studio of the Catholic University of Uruguay

Leonardo Fiorelli
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Abstract
The comparative results of the pre- and post-construction acoustic assessment of the TV Studio of the Catholic University of Uruguay are presented in this paper. The test proofs are also described. An acoustical diagnosis of the room where the studio should be built was done at first. Based on these results, a Target Curve was defined for developing the preliminary designs of the TV Studio, and this Target Curve allowed to obtain the Project Curve. During the building process, many absorbent / reflective panels were designed and built in situ to achieve the target reverberation time. Some other restrictions should be

satisfied, as using only fire-retardant materials. Closely accompanying the building phase was the key for achieving the target values of the main acoustic parameters as prescribed in the Project. When the construction phase was over, users have highly appreciated the improvement of the acoustic performance of the Studio where they perform their academic tasks. Even though, another set of acoustic tests were performed to objectively compare the target curve and the project curve of the room. All the acoustic tests were performed with two different acoustic signals: electroacoustic frequency sweep and a whip. In both cases, the plug-in Aurora by Angelo Farina for audio editing software Audacity was used.

Acoustics in Education:

Paper ICA2016-817

Speech transmission index variation due to ventilation and air-conditioning system in university classrooms

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Abstract

A major factor to be considered in the study of teaching-learning process based on spoken communication is sound field. Excessive background noise levels or reverberation time values inside a classroom interfere with oral communication, creating an acoustic barrier to the teaching-learning process. In previous work, reverberation time was measured in a sample of university classrooms in the City of Córdoba, observing that such parameter has tripled the value recommended by international literature. Furthermore, background noise level was measured in two different conditions: heating, ventilation and air-conditioning system turned on and off. Then, these levels were evaluated according to NC and RC criteria established by IRAM 4070, where more than 50 % of the population under study complied with the NC criteria, and none with the RC criteria. Moreover, spectrum of heating, ventilation and air-conditioning system noise was observed to significantly interfere with the frequency range with greater contribution to speech intelligibility. In this paper, results of speech transmission index (STI) measurements for both conditions of heating, ventilation and air-conditioning system are presented. Such measurements were performed by the impulse response method according to IEC 60268-16, with head and torso simulator as source. Results of STI and signal-to-noise ratio obtained in both conditions are compared. Variations of up to twelve times the just-noticeable difference of STI were found.

Acoustics in Education:

Paper ICA2016-102

Global case studies of acoustics in classrooms

Kenneth P. Roy

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Abstract

Classroom acoustic standards and guidelines are being discussed around the world now that the design community is becoming well aware of the need for good acoustic performance in schools, and especially in grades K-12. Teachers teach and the students learn primarily by hearing and seeing the teachers work. School designers are aware that 3 aspects of the building design are important in providing an effective acoustic environment, these being 1) a high level of speech clarity, 2) adequate signal-to-noise ratio above the ambient mechanical/electrical/plumbing noise, and 3) sufficient blocking of activity noise from adjacent spaces. Speech clarity is addressed by the architectural design, noise by the ambient noise level from building systems, and activity noise intrusion by the barriers provided in the wall/ceiling systems. Schools in both North and South America and in China and India have been evaluated to date. Evidence based design studies have been conducted, and those results including measurement data and audio/video samples are being presented. In each case, an architectural intervention was conducted between the before/after evaluations that include both objective measures of performance and subjective perceptions by the students and staff. Good design is both noticed and valued by both the students and the teaching staff.

Acoustics in Education:

Paper ICA2016-268

Intelligibility in a classroom: Coupling effects between outdoor noise and room acoustic response

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Abstract

Background noise level and reverberation time are the standard parameters used to evaluate speech understanding in a classroom. This space can be excited by an outdoor source of noise even when it has an acoustic treatment. The room geometry, usually rectangular, creates differences in the signal-to-noise ratio S/N between all listening positions due to eigenmodes. In the present study, 5 listening positions are used in a standard size classroom excited by a virtual source of pink noise. This source is placed in a corner to represent an external noise source. Using the matrix sentence method developed by Hochmuth et al. [Int. J. of Audiology, 51, 536–544 (2012)], 300 sentences from a virtual speaker are used as the signal source located at the opposite corner. Speech and noise sources are adjusted to a L_{Aeq} of 65 dBA in the reference point at the center of the room. Level difference among the positions is less than 6 dB considering the A-weighted S/N (S/N(A)) but the S/N is higher than 10 dB. Both sources are recorded simultaneously with a dummy head in all positions. A subjective test is then performed with headphones to evaluate intelligibility in these listening points. Room acoustic parameters for intelligibility, D_{50}, STI, S/N and S/N (A), are compared with the subjective test results. Results show that speech recognition is different between all points but no correlation can be found with the objective parameters. In this context, intelligibility is not being affected by room eigenmodes.

Wednesday afternoon, 7 September 2016 — **Room 204**
14:30 - 16:10
Physical Acoustics
PA3 - Ultrasound

Ultrasound:

Paper ICA2016-305

Measurement of dynamic Young's modulus by ultrasonic resonance with cylindrical rods and finite element modelling analysis

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Abstract

This work provides first measurements of a method to obtain the dynamic elastic modulus from the fundamental frequency of the longitudinal mode of a cylindrical rod, using a sonotrode (piezoelectric transducer with mechanical amplifier) as the signal source. The transducer can generate longitudinal mechanical waves from an electrical signal, in particular a sine sweep. To perform this measurement, the transducer is characterized with laboratory tests and simulation by finite element modelling software with COMSOL®. This paper seeks to link this technique with the metallurgical industry and geology, collecting literature that justifies the need on developing faster, portable and inexpensive techniques to achieve measuring the dynamic Young´s modulus of materials with reasonable accuracy.

Ultrasound:

[Paper ICA2016-386]

Type IV composite pressure vessel characterization by measurement of acoustic reverberation

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Abstract

Vehicles' CO2 emissions are being reduced to adapt to climate change. Today, the primary energy source of CO2 free vehicles use hydrogen stored in type IV Composite Overwrapped Pressure Vessels (COPV). Different non-destructive testing (NDT) methods exist to follow vessel's integrity during its operating life. The propagation of acoustic waves in a finite medium with low attenuation produces long duration signals (reverberation). These complex signals contain useful quantitative and qualitative information about medium properties and are sensitive to structural changes. Available proper data extraction techniques could thus be convenient for structural characterization. A well known example in room acoustics is that reverberationtime (RT) is directly related to walls absorption. Yet, this method is not commonly used in NDT of solid media. A type IV COPV consists of at least one aluminum base, a polymer liner and acomposite structural shell. In this work, we aim to characterize the aluminum-polymer interface adherence. We adapted Sabine's concept (room acoustics) by establishing a relation between the RT and solid interfaces'absorption. Experiments were conducted on aluminum-base free and aluminum-polymer with different bonding conditions. Experimental setup consists on asetof 9 piezoelectric (PZT) patches placed on the aluminum-base surface. One PZT patch is used as acoustic source while the others work as receivers. Then, the RT of the linear-system is obtained from the Schroeder's integrals average over the received signals, absorption is estimated and adhesion level is deduced. Our experimental results show the potential of this method to distinguish different bonding conditions and provide a good alternative for NDT of the COPV's interfaces' operating life.

Ultrasound:

[Paper ICA2016-531]

Analytical solution for modal acoustic propagation with laminar mean flow in 2D Cartesian geometry

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Abstract

An analytical for the propagation of sound in 2D Cartesian geometry in the presence of laminar mean flow is derived. This solution, using Kummer's formalism generalizes previous results obtained in uniform flow and the case without flow. The dispersion equation is then established in the case of hard-wall conditions. Solving this equation allows the calculation of propagation constant, allowing in turn the calculus of acoustic pressure profiles. In a second step, connections between the derived solution and more simple solutions (uniform flow and no flow case) are made for the acoustic pressure profiles but also for the dispersion equation. The second part of the proposed communication is dedicated to the analysis of the evolution of propagation constant and acoustic pressure profile with Mach number and working frequency. This analysis is performed for downstream and upstream configurations. Evolution of cut-off frequency against Mach number is also discussed. The dispersion map for the first modes is then plotted and concludes the communication.

Ultrasound:

Paper ICA2016-741

An ultrasonic waveguide sensor for monitoring alcohol concentration in water-alcohol mixtures

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Abstract

In this study, a sensor system that can measure the alcohol concentration of a mixture composed of ethanol and water using guided ultrasonic waves is presented. The complexity of the problem arises from the fact that ethanol-water mixtures can constitute acoustic velocities higher than the individual acoustic velocities of water and ethanol. A measured acoustic velocity may correspond to two different compositions, requiring both acoustic velocity and density of the mixture to be measured simultaneously to identify the alcohol concentration of the mixture. The sensor system, utilizing Scholte waves, enables measurements inside a closed container through a waveguide, without the mixture exposed to air. Following a detailed dispersion analysis, Scholte waves are employed using mode conversion from Lamb waves utilizing a piezoelectric transducer attached to a thin wave-guiding plate to measure the acoustic velocity and density of such mixtures.

Ultrasound:

Paper ICA2016-943

A theoretical analysis of acoustic jets

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Abstract

In this work, we report for the first time on the formation of acoustic jets (acoustojet) in the plane wave scattering by a sphere of several wavelengths in diameter. This phenomenon is the acoustic analogue of photonic jet. An acoustic jet is a narrow and high-intensity beam that emerges from the shadow-side surface of the sphere. A striking feature of these jets is the possibility of focusing beyond the diffraction limit, which restricts beamwidth be larger than the wavelength l. If the incident wave is focused on the sphere's surface at the shadow's side, the acoustic jet width is smaller than the wavelength and reaches a high intensity. According to geometric optics approximation this happens when the refractive index contrast is 2. In our analysis, we use the rigorous partial-wave expansion method to obtain the scattered pressure around the sphere. In case of a solid elastic sphere both compressional and shear waves are taken into account. Investigations show that for a 5l-radius sphere with refractive index of about 1.6, the acoustic jet remains under the diffraction limit by approximately few wavelengths in depth, with an intensity gain close to 20dB referenced to the incident intensity. Several examples of acoustic jet by a homogeneous sphere made of polyethylene, silver, and lead are illustrated and characterized according to beam waist, intensity gain, and propagation depth.

Underwater Acoustics:
Paper ICA2016-262

Application of wavelet transform for classification of underwater acoustic signals

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Abstract
Inspired by the experience of training human experts in sonar, automatic classification of signals detected by sonar is used to recognize the platforms. Many techniques of feature extraction have been developed, such as Mel-frequency cepstrum coefficients, to simulate passive target signal. The paper proposes a method that integrates wavelet transform to the feature extraction method, and the resulting features are employed for the classification problem. The classifier identification rate is calculated, and the performance of the recognition model is evaluated for different range, speed, and direction for the maritime target. Moreover, the performance of the classifier in noisy conditions is investigated.

Underwater Acoustics:
Paper ICA2016-398

Using base function decomposition in adaptive filtering for seismic pulse retrieving

Marina Bousfield[(a)], João Ristow[(b)], Julio Cordioli[(a)], Izabelle Aller[(a)], Marcus Barão[(a)], Antonio Klein[(a)], Guillaume Barrault[(a)]
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Abstract
Concerning the investigation of crustal structure below the water column, seismic processing is still among the most used techniques. A crucial stage in seismic data processing is the deconvolution between seismic trace and the pulse emitted by the source, which allows estimating the reflectivity of the studied substrate. This operation has known sources of errors caused mainly by: (1) the absence of a priori knowledge of the seismic pulse shape; and (2) the deconvolution algorithm used. This work proposes a method to obtain the shape of the pulse emitted to the environment using recorded seismic data, aiming to improve the deconvolution process. The waveform of the seismic pulse can be represented as a combination of simple functions, like sine and exponential functions. To obtain the functions coefficients, an adaptive filter feed by a Least Mean Square (LMS) algorithm is used. The filter's adaptive coefficients are updated in a feedback loop creating an estimator that reconstructs the original waveform. Several pseudorandom waveforms were empirically created and the presented algorithm was used to retrieve its shapes. As a result of reduction of the degrees of freedom caused by the decomposition in base functions, the described method proved to be faster and robust comparing with other ones. In the future, the proposed processing will be used to deconvolve real seismic data.

Underwater Acoustics:
[Paper ICA2016-35]

Numerical and experimental prediction methods of cavitation noise radiated by underwater propellers

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Abstract
Underwater propeller cavitation noise is composed of tonal blade rate noise and high frequency broadband noise. In this paper a numerical method is developed to predict propeller tonal noise while experimental approaches are performed to predict broadband noise. For prediction of the sheet cavitation which contributes to tonal noise characteristics, its area and volume on the propeller blades are calculated by finite volume method. Then propeller tonal noise is calculated using the acoustic analogy with consideration of cavitation volume variation on the blade surface in the time domain. This procedure was validated with the acoustic measurement test in the water tunnel. The experimental approaches for propeller broadband noise are composed of the development of semi-empirical formula through water tunnel test and the onboard measurement in the real ship. The semi-empirical formula for tip vortex cavitation noise is developed based on the aero-acoustic theory of tip vortex formation noise and then optimized through water tunnel test for nine kinds of model propellers. The transfer function is developed to acquire quasi-free field acoustic response using an underwater loudspeaker in the water tunnel and the towing tank. The propeller broadband noise is also predicted by sound transmission coefficient method using the relationship between sound transmission coefficients in the dry dock and strucutre-borne noise measurement at the sea trial. The proposed methods were validated by underwater radiated noise measurement during sea trials. From the results, it is expected that the proposed methods enable to predict propeller cavitation noise with ease and accuracy.

Underwater Acoustics:
[Paper ICA2016-164]

Numerical study of unsteady supercavitation and flow induced noise propagation

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Abstract
Operation of hydraulic devices and underwater vehicles is often accompanied by occurrence of flow cavitation/supercavitation. Due to the transient and unstable nature of cavitation phenomenon, it is difficult to predict the evolution of unsteady cavitation/supercavitation. The current study is focused on the numerical investigation of supercavitating liquid flow perturbed by pressure wave with one-fluid cavitation model. It is observed that the supercavity will become unstable under the impact of pressure wave and may collapse or deform locally, depending on strength of perturbation. Collapse of supercavitation results in huge pressure surge, which may cause the material erosion, noise, vibration and efficiency loss of operating devices. Understanding of unsteady supercavitation is crucial for design and optimization of such devices. Separately, the supercavity inception/development and its sustainment through ventilated cavitation may result in turbulence and fluctuations at the water-vapor interface, which are major sources of hydrodynamic noise. Here, three main sources are investigated: flow generated noise due to turbulent pressure fluctuations around the supercavity, pressure fluctuations at the vapor-water interface and pressure fluctuations due to direct impingement of ventilated gas-jets. In this study, the near and far field noise contributions from each of the aforementioned sound sources are considered via the boundary

integral method. BEM based acoustic solver was developed for computing flow generated sound with flow data obtained from the CFD solver for simulating supercavitation flow.

Underwater Acoustics:

[Paper ICA2016-446]

Exploitation of frequency information in Continuous Active Sonar

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Abstract

In pulsed active sonar, short duration coded waveforms insonify the area of interest. The low duty cycle limits detection opportunities and decreases average energy. A recent concept is continuous active sonar (CAS), which has continuous source transmission over a broad frequency band. Previous work by the authors has investigated the utility of extracting the propagation-induced frequency structure in pulsed sonar. The broadband, continuous CAS waveforms particularly lend themselves to this approach. The presence of active striations in CAS data has been recently identified in the shallow water Target and Reverberation Experiment (TREX13). In this paper we provide additional examples of frequency structure in both the TREX13 and simulated data, and discuss methods for exploiting the striations to improve tracking performance. In pulsed active sonar, short duration coded waveforms insonify the area of interest. The low duty cycle limits detection opportunities and decreases average energy.

Wednesday afternoon, 6 September 2016 **Lounge Lateral Room**
14:50 - 16:10
POSTER SESSION - Monitor 1
Architectural Acoustics - Room and Building Acoustics
AA6 - Concert Hall Acoustics

POSTER

Concert Hall Acoustics:

[Paper ICA2016-676]

Analysis of reverberation times and energy decay curves of 1/12 octave bands in performance spaces considering musical scale

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Abstract

Generally in the room acoustic design field, the reverberation times of 1 octave bands or 1/3 octave bands are analyzed for evaluating acoustic conditions in performance spaces such as concert halls. The center frequency in the octave bands is defined by engineering frequency intervals such as 250 Hz, 500 Hz, and 1000 Hz in the conventional analysis method. However, this definition of center frequency is not always adequate for evaluating performance spaces because music is composed of musical notes based on a musical scale, which is different from engineering frequency scales. Therefore, we studied the analysis of reverberation times and energy decay curves in 1/12 octave bands based on an equal tempered scale. Reverberation times in several concert halls were analyzed by this method. Distributions of reverberation times of 1/12 octave bands between concert halls were obtained. Next, instrumental sounds of varying pitches were convolved with impulse responses, and the reverberation times of the different pitches were analyzed from the convolved signals. As a result, variation of the reverberation times of the 1/12 octave bands and that of the different pitches corresponded well. When people listen to music in a performance space, they feel some impressions of the music, of which chords are an important aspect. Thus, chord signals were convolved with impulse responses, and the convolved signals of the reverberation times were analyzed. Finally, distributions of the differences of energy decay curves of chord signals were obtained.

POSTER

Concert Hall Acoustics:

Paper ICA2016-511

Acoustics of new and renovated chamber music halls in Russia

Alexander Fadeev, Nikolay Kanev, Anatoly Livshits, Andrey Nechaev, Anton Peretokin, Eugeny Pimenov, Vitaly Rodenkov, Natalia Shirgina

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Abstract

In recent years several classic halls for chamber music were renovated and a few halls of similar capacity were built in Russia. We have selected the most interesting ones and measured their acoustical parameters. Here we are presenting detailed descriptions of five halls in Moscow, Tula, Ufa and Penza in combination with the measurement data. Some of them have long reverberation time and their acoustics is considered as very good. In two halls there is an organ, else one is going to be supplied with new organ. Moreover, the hall in Ufa is destined not only for chamber music; amplified performances may take place there. In spite of big difference between these halls we try to find connecting features and compare them with well-known chamber music halls. One of the main results is that we revealed the trend that general audience has subjective preferences towards longer reverberation time. Those results were obtained at the same time as evaluation of acoustics changes in Bolshoi Theatre Historic stage and Great Hall of Moscow Conservatory due to reconstruction. Else one interesting result of reverberation time measurement in occupied and unoccupied hall is presented in the paper.

POSTER

Concert Hall Acoustics:

Paper ICA2016-502

Rank-ordering opera houses according to their sound quality parameters using PROMETHEE II method

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Abstract

Classifying sound quality in opera houses is a difficult task, as it depends on several subjective and objective parameters. The aim of this work is to create and evaluate a method for objective rank-ordering of Opera Houses and to compare it with the subjective rank-ordering procedure proposed in Hidaka and Beranek, J. Acoust. Soc. Am. 107, 368-383 (2000). This problem was solved using a multicriteria decision making method, called PROMETHEE II. This article describes its implementation and presents experimental results which are compared with subjective results presented in Hidaka and Beranek (2000). In addition, PROMETHEE II was used to classify 9 opera houses which had not been ranked in Hidaka and Beranek (2000). The results obtained show that the objective rank ordering created using PROMETHEE II which is based on objective parameters can successfully replace the subjectively based procedure of Hidaka and Beranek (2000). Four of the top six ranked opera houses were correctly classified and the performance can still be improved by means of optimization methods currently under investigation.

POSTER

Concert Hall Acoustics:

Paper ICA2016-694

Influence of music-induced floor vibration on impression of music in concert halls

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Abstract

Sound can be perceived in two ways: using the auditory sense, and as vibrations, using the tactile sense. Several musicians and music-lovers recognize that the floor and audience seats vibrate during chamber music performed in concert halls. There are several studies on the psychoacoustic role of vibration in the field of room acoustics. Most recent studies have investigated only the vibration generated on the stage, but there are a few studies on the vibration of the audience floor. Depending on the structure and materials used in construction, the vibrations caused by the musical performances may become sufficiently strong to be perceived by the audience, which, it can be supposed, may affect the auditory impression of music. In order to confirm this hypothesis, an apparatus was prepared to simulate the vibration of the audience floor. This study discusses mainly two experiments using the apparatus. First, subjects listened to music with or without vibration synchronized with the music using the apparatus. After listening to the music, the Evaluation Grid Method was used to extract words of impression. Second, we investigated whether the impressions of the music were influenced by the magnitude of vibration. Additionally, in order to confirm whether vibration really occurs in real concert halls, vibrations were measured in the concert halls.

Wednesday afternoon, 6 September 2016 — **Lounge Lateral Room**

14:50 - 15:10

POSTER SESSION - Monitor 2

Acoustical Oceanography

AO1 - Acoustical Oceanography

POSTER

Acoustical Oceanography:

Paper ICA2016-518

Concurrent acoustical and optical observations in zooplankton studies

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Abstract

Zooplankton is an important component of the marine ecosystem linking primary producers with higher trophic levels. It is characterized by irregular distribution in the water column, forming aggregations called patches. It is difficult to collect complete information on zooplankton distribution with traditional methods (e.g. nets), that provide discrete and low resolution data on distribution of zooplankton biomass and abundance, therefore use of alternative methods is necessary. Acoustic sounding makes environmental studies fast, non-intrusive, and relatively cheap with high temporal and spatial resolution. Laser Optical Plankton Counter (LOPC) delivers real-time information on zooplankton abundance and size spectra. Presented results are based on the data collected in the Gulf of Gdansk, southern Baltic Sea collected during cruises in the years 2013-2014. Data was collected during simultaneous profiling with high frequency (420 kHz) echosounder and LOPC tows along the chosen transects. Zooplankton size spectra obtained by LOPC were used as input parameters in mathematical model of sound scattering on zooplankton individuals. Model output values of acoustic backscattering strength were then compared with values obtained by echosounding. Zooplankton size structure is helpful in validating and refinement of sound scattering model for specific set of scatterers. Acoustical measurements supplemented by optical observations have a great potential to describe zooplankton distribution over large scales.

POSTER

Underwater Acoustics:

Paper ICA2016-151

A study on spatial fading in a shallow water channel

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Abstract

A methodology to assess the spatial fading suffered by acoustic signals transmitted in a Underwater Acoustic (UWA) channel in Arraial do Cabo, Brazil, is provided. This study was necessary in order to predict how the nominal conditions of this shallow water environment impose spatial variations on the Transmission Loss (TL) of a propagating signal. A Ray Tracing model was used to predict the underwater acoustic wave field (frequency range from 5 kHz to 10 kHz). The numerical model takes into account the geometric and geo-acoustical properties of the bathymetry founded at the studied area. In addition, the influence of two distinct range dependent Sound Speed Profiles (SSP) on spatial fading is investigated. One SSP was measured during a coastal upwelling event and the other in regular sea conditions. The result is a methodology of analysis that not only estimates the TL due to geometrical spreading, sound absorption and multipath propagation, but also the signals intensity fluctuations along the channels range. Thus, this study allows a better understanding about the signal's behavior on this environment aiming the use of diversity techniques to lower the effects of signal fluctuations on a receiving system.

POSTER

Underwater Acoustics:

Paper ICA2016-192

A new improvement of RAKE algorithm based on prediction of multipath structure of underwater acoustic channel

Zhi Shaolong [a], Wang Yongfeng [a], Zhang hong [a], Che hong [a]

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Abstract

An underwater acoustic spread-spectrum communication system is established in this paper. Compare the performances of a direct sequence spread spectrum system and a Rake receive system through simulation based on the decision of multipath structure under the assumption of dual beam channel in shallow water with thermocline, the approximate expression of the bit error rate is given. Simulation results show that multipath structure with certain features could truly play the role to eliminate inter-symbol interference (ISI) and improve performance of the receive system. A demodulation algorithm is proposed based on the characteristics of multipath structure prediction and the feasibility of it is verified through processing the trail data.

POSTER

Underwater Acoustics:

[Paper ICA2016-235]

A Survey on recent researches of MAC protocols in UASNs

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Abstract

With the developments of sensor technology and signal processing technology, it becomes feasible to construct a large-scale underwater acoustic sensor network (UASN), while the propagation speed in underwater acoustic is 5 orders of magnitudes lower than that in wireless channel, along with the huge influence of the complex underwater ambient noise, which results to the bad performance of UASNs. Furthermore, UASNs nodes are powered by battery, the energy-limited nodes are highly difficult to deploy and recycle. These cases make transmission validity and energy efficiency important factors to evaluate an UASN. MAC protocol plays a critical role in UASNs, determining the network performance in some degree, which draws attention of many researchers and institutions, they have continuously proposed some MAC protocols fit for underwater acoustic environment, and verified their performances in simulations and real experiments. This paper focus on the excellent researches in a decade, a detailed survey about the progress of the advanced technology will be made, besides there is an illustration about its future direction.

Wednesday afternoon, 6 September 2016 **Lounge Lateral Room**
14:50 - 16:10
POSTER SESSION - Monitor 3
Speech Communication
SC1 - Speech Communication

POSTER

Speech Communication:

[Paper ICA2016-645]

An acoustic examination of laterals in Lower Sorbian

Phil Howson[a] Alexei Kochetov[b]
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Abstract

This study is an acoustic examination of laterals in Lower Sorbian, an endangered language spoken in eastern Germany. Gestural timing effects of the English lateral, /l/, have been well studied, and overall the data suggests different timing patterns for the tongue tip and tongue body gestures in syllable initial (‘clear l’) and syllable final positions (‘dark l’). The coordination between the tongue tip and body in dark laterals has been used to suggest that misperception has led to the cross-linguistically common change from dark /l/ to /w/. Acoustic data for two dialects of Lower Sorbian (one with the dark lateral and one which underwent the sound change to /w/; both have a clear lateral, /l/) was collected and analysed using an Smoothing Spline ANOVA to examine temporal characteristics of these sounds.The findings suggest that there is a difference in the gestural timing between the clear and dark lateral in word initial and final positions. The dark lateral has earlier achievement F2-F1 target, suggesting an earlier achievement of the tongue dorsum retraction; F2-F1 remains low throughout the following vowel, suggesting a backing effect. The clear lateral has intensifying F2-F1 suggesting tongue dorsum retraction over the duration, until the onset of the following vowel. The data also suggests that lenition of the tongue tip in the word final position could likely have been a trigger for the sound change. In word final position, there is also a much higher F3, suggesting weaker lip rounding, resulting in a similar F3 frequency and trajectory for the dark lateral and /w/. This also suggests a similar vocal tract shape between the dark lateral and /w/. Both that dark lateral and /w/ also show a lowering effect on the F2 of the vowel following them.

POSTER

Speech Communication:

Paper ICA2016-140

Context speech rate can affect Arabic geminate perception

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Abstract

The influence of temporally-removed (i.e., distal) speech rate information on word segmentation has been a recent topic of interest in the field of phonetics. However, cross-linguistic verification of distal speech rate effects is still fairly uncommon, a particularly glaring oversight given the potential relevance of cross-linguistic studies to important unresolved questions. Here, we look at distal speech rate effects on the perception of consonant length (rather than word segmentation) in Modern Standard Arabic. We find that distal rate effects are also present for the perception of geminate consonants, which suggests that distal speech rate effects are capable of being extended to the perception of segmental contrasts as well as word segmentation.

POSTER

Speech Communication:

Paper ICA2016-544

The acoustics of word-final fake gemination in Egyptian Arabic

Kyle S. Jones

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Abstract

Holes (2004) observes that neutralization of word-final geminates (long consonants) in some dialects of Arabic ([sakat] “he was quiet” vs. [sa kat-t] “I was/you (fem.) were quiet”) may lead to phonemic stress if degemination occurs (e.g. [sa kat]) (62). I explore this understudied issue as it concerns Egyptian Arabic; analysis of acoustic data demonstrates that degemination does not occur, speakers maintaining acoustic distinctions between word-final singletons and word-final fake geminates, which result from proximity of identical consonants. Three speakers participated in the present study, reading fifteen verbs in two different forms from a list. Two speakers from Alexandria read 3MSG imperfect and 2SG imperfect forms, and a third speaker from Qalubiya read the 3MSG imperfect and 2FSG imperfect for comparison between word-final and intervocalic geminates. The recordings were measured in the phonetics program Praat (Boermsa & Weenink 2015) for preceding vowel length, stop closure duration, and length of stop burst. Word-final fake geminates were found to be 1.3 times as long as singleton consonants, with no appreciable difference in vowel length preceding singletons and geminates. The intervocalic “geminates,” however, were found to have a 1:1 ratio to their singleton counterparts. These findings are contrary to research on word-final true geminates in Urban Jordanian Arabic (Al-Tamimi, Abu-Abbas, & Tarawnah 2010), which found that word-final true geminates were 1.5 times longer than singleton consonants, with shorter preceding vowels. This may suggest that true and fake geminates in Arabic varieties are distinguished by their differing acoustics. I conclude with a typology of strategies that varieties of Arabic and other languages may use to preserve fake geminate contrasts.

POSTER

Speech Communication:

Paper ICA2016-770

Effects of accent on cortical auditory evoked potentials for vowel contrasts

Emma Brint[a], Petra Hödl[b], Paul Iverson[c]

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Abstract

Accent differences affect speech recognition, with listeners finding it easier to understand speech spoken with an accent like their own, and more difficulty when the accent is unfamiliar. Most work has examined the role of accent in higher-level linguistic processing (e.g., phonetic categories), but the present study investigated how the listener's accent influences early auditory-cortical responses for vowel contrasts. Listeners with different English accents (southern British English, Glaswegian, native Spanish speakers) heard random series of concatenated southern British English vowels, and EEG was used to record the cortical auditory evoked potentials in response to each vowel spectral change. There were different patterns of responses for the listener groups, with the non-native English listeners (Spanish) having smaller responses overall to English vowels, and Glaswegian listeners having smaller responses than Southerners only for vowel contrasts that are realised differently between the two accents (e.g., *beat-bit*). The results thus add to the evidence that language experience can have effects at early levels of auditory processing rather than only affecting linguistic categorization.

Wednesday afternoon, 7 September 2016 **Cardenal Pironio Auditorium**

16:30 -18:30

ICA GENERAL ASSEMBLY

Thursday, 8 September 2016

Thursday morning, 8 September 2016 **Juan Pablo II Auditorium**
09:00 -10:20
Environmental Acoustics & Community Noise
EN4 - Smart City Sound Monitoring

INVITED

Smart City Sound Monitoring:

[Paper ICA2016-97]

Robust microphone array beamforming for long-term monitoring of industrial areas

Bert De Coensel[a, b], Dick Botteldooren[a], Timothy Van Renterghem[a], Luc Dekoninck[a], Vincent Spruytte[b], Alphonso F. Makovec[c], Peter Wessels[d], Frits van der Eerden[d], Tom Basten[d]

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Abstract

Noise generated by industrial sources is often a cause of annoyance for people living in the surroundings of industrial areas, even when all individual plants adhere to regulations and possess all mandatory environmental permits. The variety of industrial noise sources is wide and the noise generated can be complex, making noise monitoring of large industrial areas not trivial. Low-frequency noise sources in particular are hard to localize using conventional sound measurement equipment, and may give rise to noise annoyance complaints at great distance from the source. This paper reports on the development of a 50m-wide robust microphone array system designed for long-term monitoring of industrial areas. The microphone array system consists of a set of curvilinearly and non-equidistantly spaced microphones, allowing to cover a broad spectral range and to minimize front-back confusions. The audio signals recorded by the microphones are periodically synchronized using a test signal played back through a set of two loudspeakers that are placed at each end of the array. This synchronization method also produces an estimate of the local, instantaneous speed of sound, which serves as a validation of the synchronization process, and which is used within the free-field beamforming algorithm. Finally, a high-end microphone unit is placed near the array to achieve accurate sound power level estimates. Field tests conducted in strong wind conditions with the microphone array demonstrate its effectiveness in detecting the direction of various types and combinations of sound sources with good angular resolution.

Smart City Sound Monitoring:

[Paper ICA2016-330]

Removing local sound disturbances from industrial noise monitoring at long distance

Peter Wessels[a], Frits van der Eerden[a], Tom Basten[a], Bert de Coensel[b,c], Dick Botteldooren[c], Timothy van Renterghem[c], Luc Dekoninck[c], Vincent Spruytte[b], Alphonso Makovec[d]

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Abstract

Industrial areas with heavy industry may cause annoyance for neighboring residential areas. Especially sources that emit low frequency noise can cause annoyance in residential areas several kilometers from the industry. At these distances the low frequency sound propagation is largely

dependent on the meteorological conditions. Height dependent wind and temperature profiles result in a varying effective sound speed over altitude, causing upward or downward sound paths. Depending on the meteorological conditions this can have a significant effect on the acoustic transfer from source to receiver. This paper presents a method to estimate the acoustic immission caused by heavy industry. Acoustic data is collected by monitoring of an actual industrial area, over long distance and over a time period of multiple months. A meteorological acoustic transfer model combined with measurement based source emission estimations is used to estimate the immission of the industrial sources in a residential area. Transfer distances range from 2 km up to 10 km. Together with the emission of the industrial sources also the acoustic immission is continuously measured by acoustic monitoring stations within the residential area. The calculated immission estimates are used to differentiate between the industrial related noise and all other sounds within the residential area. When considering possible annoyance due to the industry, the monitoring of other sounds is not relevant and should be excluded from the analysis. This differentiation is made by comparing the estimated immission with the measured sound level for the residential area. The method is an alternative for the audio classification methods of industrial or other sounds.

Smart City Sound Monitoring:

Paper ICA2016-545

A proposal for a mobile phone´s application alerting and warning about noise pollution

Verónica Arroyo-Pedroza[a], Fausto E. Rodriguez-Manzo[b],
Roberto A. García-Madrid [c], Iván Diaz-de-León[d], Héctor Reyes Aguilar[e])
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Abstract

The emerging concept of Smart Cities promotes self-managing citizen relationships: a way of improving efficiency around resources. An increasing number of cities currently opened to free Wi-Fi policy look for greater democratization of smartphones. Apps market is rapidly growing and widening promoting social interaction and social activism. Noise apps like NoiseWatch, NoiseTube, or those like sound level meters allow collecting data and sending it by e-mail. In order to promote public awareness about noise pollution in Mexico City's Metropolitan Area, we present here a proposal for a cell-phone application bound to collecting data about citizen's concerns regarding environmental noise. The aim of our project is to develop an application that will connect with citizens from a friendly, practical and accesible interface, inviting them to collaborate and benefit themselves and their community by alerting and raising awareness about the importance of environmental noise and its effects on health. We envision a tool that helps, educate, alert and link people to take decisions on this important pollution problem. A Mexico City Metropolitan Area Noise Map has been already made, showing the impact of noise levels emitted by road traffic noise on main city roads. However there is a need to develop other tools which can interact with citizens giving them the possibility of developing a digital map of Mexico City that can display the status of noise in it, in real time. This paper presents an approach to that tool: the architecture and content of the app, along with an interface that will seek to promote fundamental citizen's participation and the kind of results that are expected.

Smart City Sound Monitoring:
Paper ICA2016-793
Data mining on urban sound sensor networks
Dick Botteldooren[a], Talis Vertriest[a], Michiel Boes[a], Bert De Coensel[a], Pierre Aumond[b], Arnaud Can[c], Carlos Ribeiro[d], Catherine Lavandier[b]
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Abstract
Urban sound sensor networks deliver megabytes of data on a daily basis so the question on how to extract useful knowledge from this overwhelming dataset is eminent. This paper presents and compares two extremely different approaches. The first approach uses as much as possible expert knowledge on how people perceive the sonic environment, the second approach simply considers the spectra obtained every time step as meaningless numbers yet tries to structure them in a meaningful way. The approach based on expert knowledge starts by extracting features that a human listener might use to detect salient sounds and to recognize these sounds. These features are then fed to a recurrent neural network that learns in an unsupervised way to structure and group these features based on co-occurrence and typical sequences. The network is constructed to mimic human auditory processing and includes inhibition and adaptation processes. The outcome of this network is the activation of a set of several hundred neurons. The second approach collects a sequence of one minute of sound spectra (1/8 second time step) and summarizes it using Gaussian mixture models in the frequency-amplitude space. Mean and standard deviation of the set of Gaussians are used for further analysis. In both cases, the outcome is clustered to analyze similarities over space and time as well as to detect outliers. Both approaches are applied on a dataset obtained from 25 measurement nodes during approximately one and a half year in Paris, France. Although the approach based on expert knowledge is expected to be much more precise when it comes to analyzing and clustering soundscapes, it is also much slower than the blind data analysis.

Thursday morning, 8 September 2016 **Juan Pablo II Auditorium**
11:00 -12:00
Environmental Acoustics & Community Noise
EN3 - Road Traffic Noise Modeling and Noise Barrier

Road Traffic Noise Modeling and Noise Barrier:
Paper ICA2016-850
A field survey on sound power level of Japanese road vehicles
Shinichi Sakamoto[a], Hyojin Lee[b], Miki Yonemura[c]
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Abstract
For precise prediction of road traffic noise, accurate sound power levels of road vehicles are necessary as a fundamental input to a calculation of road traffic noise. In Japan, the road traffic noise prediction model “ASJ RTN-Model 2013”, which was proposed by the Acoustical Society of Japan, is usually used for assessment of road traffic noise. In the model, sound power levels of vehicles are given as simple equations and the parameter values included in the equations are determined for some vehicle categories. Recently, hybrid and electric vehicles are increasing especially in the category of passenger car. In addition, light vehicles with a cubic capacity of 660 cc. or less are also increasing. Such vehicles have less sound power than gasoline engine vehicles in general, and therefore the parameter values in the model may not correctly reflect current situation. To grasp sound power levels of current road vehicles, the authors conducted field measurements on sound power levels of actual vehicles at some measurement sites. The measured sound power levels were arranged as a function of running speed and differences of the sound power levels among gasoline engine passenger cars, hybrid vehicles and light vehicles were quantitatively discussed.

Road Traffic Noise Modelling and Noise Barrier:

Paper ICA2016-564

A study of the acoustics of the urban space in Mexico City through physical scale models: A first approach

Fausto E. Rodríguez-Manzo[a], Laura Lancón-Rivera[a], Elisa Garay-Vargas[a], Ernesto Vázquez-Cerón[b], Jorge Chávez-Gómez[b], Silvia García-Martínez[a], Rafael Villeda-Ayala[c], Dulce Ponce-Patrón[d]

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Abstract

The urban space has been acoustically analyzed through noise mapping as well as applied mathematics in computer simulation models. This involves the use of statistical data for modelling and the calculation and finding of areas influenced by noise. Both computer-simulated models and physical scale models are utilized to study the sound response of the architectural space. The difference between these two methods is that real sound is used in physical scale models while mathematics are used to simulate sound in computer models. In the field of urban acoustics it is important to know more about the use of real sound in models, as it can give us a more realistic idea of the acoustic behaviour of the space. The study of the acoustic response of the urban space is important because it allows us to understand the impact buildings have in the sound environment of that particular space. Furthermore, acoustic scale models provide a deep understanding of real sound interaction. This paper shows the process developed to find the different options of acoustic analysis of the urban space in Mexico City through the use of scale models. The process included reviewing scales, urban morphology, materials, sound sources, façade design, building dimensions and urban components, among others. The goal is to establish the most suitable method for this type of analysis of Mexico City, which has a specific set of characteristics. It is important to highlight that this study is proposed from an architectural and urbanism point of view.

Road Traffic Noise Modeling and Noise Barrier:

Paper ICA2016-904

Sonic crystal noise barrier using locally resonant scatterers

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Abstract

Sonic crystal barriers have been receiving recent interest as potential noise barriers to reduce traffic noise in certain frequency bands. Sonic crystals comprise of periodic arrangements of sound scatterers for which the simplest scatterer topology is a solid cylinder. This paper investigates the acoustic performance of a sonic crystal noise barrier using vertical cylindrical shells of finite height. Locally resonant scatterers comprising of perforated or C-shaped cylindrical shells are examined. Results for the barrier insertion loss show that attenuation in a broad band gap is generated due to destructive interference between the scattered sound waves within the periodic structure. The local resonance of the scatterers creates an additional peak in insertion loss, approximately predicted by the Helmholtz resonator frequency. For the case of the perforated cylindrical shells, the location of the resonant frequency is shown to be dependent on the number and size of the holes. When the resonant frequency due to the perforations occurs within the Bragg band gap, a significant increase in insertion loss across the band gap is shown to occur. For the case of the C-shaped cylindrical shells, the size of the opening is shown to have a significant effect on both the local resonant frequency and the band gap due to Bragg scattering.

Soundscape, Quality of Life, and Health:
Paper ICA2016-226

Application of the Swedish Soundscape-Quality Protocol in one European and three Latin-American cities

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Abstract
The Swedish Soundscape-Quality Protocol (SSQP) is a tool developed to evaluate and classify acoustic environments according to their acoustic perception by people. This protocol has been proposed by Axelsson, Nilsson and Berglund and it is based on a pleasantness-eventfulness principal component model. This model is feed by qualitative measurements of the extension of eight attributes of Soundscapes and allows one to classify the acoustic environments in four Cartesian quadrants: exciting, chaotic, monotonous, and calm. In the present work, the SSQP is applied in urban environments in four cities: Córdoba and Rosario (Argentina), Valdivia (Chile) and Lund (Sweden). The protocol was evaluated through 509 surveys conducted in 122 locations corresponding to 29 open public environments in these four cities. The evaluated environments included parks, squares, streets, avenues, pedestrian streets, cultural spaces, fountains and other recreational facilities. Results show that most urban parks and squares were classified as exciting, while a minority of them was assessed as calm. Most locations on streets and avenues were evaluated as chaotic, whereas most locations in pedestrian streets were categorized as exciting. Not a single urban environment was classified as monotonous.

INVITED

Soundscape: Quality of Life, and Health:
Paper ICA2016-233

Loci for urban soundscape planning, design and management

A. L. Brown[a] , C. J. Grimwood[b]
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Abstract
Soundscapes have a role in management of the acoustic environment of urban areas in the context of parks, historical and cultural places, and existing “quiet areas”, but also across residential precincts and other urban spaces. Soundscape planning is complementary to, not a substitute for, noise management approaches. It is generally clear where noise control principles should apply: viz. at sensitive receptors where adverse impacts and effects arise – under flight paths near airports or at the interface between industrial and residential precincts, for example. But the loci of application of soundscape management principles is less well understood. Set out here is a didactic schema of the spectrum of opportunities for planning of the soundscape of urban areas. Starting with a hypothetical distribution of sound levels across any area of interest, there will be a small proportion of that distribution with the highest sound levels. The rest of the distribution will be variously shaped – from highly positively skewed indicating that much of the area has lower sound levels, to distributions where a high proportion is at mid sound levels with a long tail of lower sound levels. The higher levels of sound are candidates for management through environmental noise control activities. The low end of the level distribution may be protected or managed as “quiet areas” to enhance urban soundscapes. This paper describes other loci for the application of soundscape

principles across this distribution. Soundscape design and management can include creative design of the built environment to achieve places of high acoustic quality; ensuring the availability of urban acoustic diversity; encouraging attention to subcriterion (noise) exposures and protection of place-defining sounds.

Soundscape, Quality of Life, and Health:
Paper ICA2016-265
Sound pleasantness evaluation of pedestrian walks in urban sound environments

Pierre Aumond[(a)(b)], Arnaud Can[(b)], Bert De Coensel[(c)], Dick Botteldooren[(c)], Carlos Ribeiro[(d)], Catherine Lavandier[(b)]
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Abstract
The health benefits of a daily physical activity, and of walking in particular, are widely acknowledged. However, walking in urban environment inevitably leads to an increased exposure to noise, which forms a drawback of choosing this transportation mode. Being able to estimate the sound pleasantness associated with an urban walk trip has many potential applications, such as informing pedestrians about the sound along their intended walk, which may help them to optimize their route choice. In the past decade, various studies have focused on characterizing and estimating the sound pleasantness perceived at specific locations, on the basis of perceptive and physical measurements. However, to estimate the sound pleasantness along an urban walking trip, an additional step is required, which consists of assessing how a pedestrian evaluates the overall pleasantness of a sound environment that varies along the walking trip. In this work, the results of two laboratory experiments and one field experiment are discussed, which were designed to assess the overall evaluation of the sound environment along an urban walk. Physical and perceptive measurements at specified positions or continuously along a series of tested routes are available, in addition to a global evaluation of the route. A comparison between the results of the three experiments provides a rich source of information to understand how the sound pleasantness of a pedestrian walk is evaluated. The main conclusion is that for short walks (of about 1 minute), a recency effect is observed, which tends to disappear when the duration of the walk increases.

INVITED
Soundscape, Quality of Life, and Health:
Paper ICA2016-340
Acoustic intervention at preschools had marginal effect on noise-levels but impacted on children´s perception and response to sound qualities

Persson Waye Kerstin[(a)], Irene van Kamp[(b)], Jeong-Lim Kim[(a)]
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Abstract
At pre-schools personnel and children are known to be exposed to high sound levels. Tiredness and sound fatigue among the personnel are reported, while less is known about how children are affected. A previously developed interview protocol (INCH) was used to study the effect of an acoustic intervention at seven preschools. Before, 61 children aged 4-6 yrs were interviewed and 59 after. A reduction of the sound level in a range between 1 to 3 dB LpAeq was measured using stationary noise levels meters. The results were analysed using Generalised Estimating Equations accounting for repeated measure of the intervention. The results showed that a change in noise levels in the dining/activity room positively

impacted on children’s perception of scraping and screeching sounds, frequency of reported tummy ache, and frequency of children reporting the teacher to speak with a raised voice. Perception of scraping sound per se, also impacted on angry reactions to scraping sounds, and children’s reporting on teachers speaking with raised voice. Although the intervention affected the noise levels only marginally, it seems to have influenced sound quality aspects related to the higher frequencies in the sound. The results are especially interesting given the new knowledge of children’s hearing.

INVITED

Soundscape, Quality of Life, and Health:

Paper ICA2016-167

Perceived soundscapes, human restoration and related health in green urban areas

Irene Van Kamp[(a)], Elise Van Kempen[(a)], Caroline Ameling[(a)] Wim Swart[(a)] Hanneke Kruize[(a)]

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Abstract

There is growing attention for the association between landscapes, green and blue space and human well-being, quality of life, and health., but most studies do not account for the moderating effects of the acoustic environment. This is partly due to a lack of relevant data. This paper presents the Dutch results of a European study (Phenotype) into the health effect of perceived soundscapes and a neighbourhood study in three cities in the Netherlands using the same approach. In both studies people were selected from neighbourhoods with varying levels of socioeconomic status and green space. By structured interview, information was gathered about availability (use and importance) of green space in the immediate environment, as well as the sound quality of favourite green areas. Data are also available about perceived restoration at favourite locations, and several health outcomes. This allowed for analysing the association between perceived soundscapes and indicators of health, while accounting for objective and subjective indicators of green. Perceived soundscape is a strong predictor of restoration at favourite green areas after adjustment for city, neighbourhood, demographics, and indicators of green space. The pattern was also found for symptoms, which are in addition predicted by social cohesion and restoration.

Thursday morning, 8 September 2016 **Dr. Valsecchi Auditorium**

11:00 - 12:00

Acoustical Measurements and Instrumentation

SI1 - Sound Intensity and Inverse Methods in Acoustics

INVITED

Sound Intensity and Inverse Methods in Acoustics:

Paper ICA2016-208

Using the irrotational part of structural intensity to identify sources of vibrational energy

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Abstract

The identification of vibrational sources can be done effectively by means of structural intensity estimates, as the structural intensity indicates the energy power flow of the flexural waves propagating through the

structure. Considering the irrotational part of the structural intensity estimate removes parts of the structural intensity field which are related to the near-field, thus showing the part of the vibrational field that propagates to the far field. The use of the irrotational part of the structural intensity results in a more clear view of the physical sources that are generating active power to the structure. A recently developed approach to calculate the irrotational part of the structural intensity from measurement data was used and results are compared with another method from literature. It is shown that the recently developed approach converges well and gives good results by virtue of the smoothening effect of the test functional series expansion employed.

INVITED

Sound Intensity and Inverse Methods in Acoustics:

Paper ICA2016-534

Characterization of a porous plate saturated with water using Bayesian inversion

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Abstract

The aim of this work is to estimate the properties of a poroelastic plate saturated by water. The estimated properties are porosity, tortuosity, viscous characteristic length, flow resistivity, solid density, bulk moduli of both the solid phase and the skeleton, and shear modulus. The characterization method consists of measuring the plate's reflection and transmission coefficients y = [R T]> and then solving the inverse problem y = H(x)+v, where H(x) is the forward model, x contains the parameters of interest and v is noise. The forward model uses the alternative 1962 Biot's formulation. Wave equations derived from this formulation are solved using the state vector formalism. Measurements are performed at normal incidence in a water tank in the ultrasonic regime. The sample is a ceramic QF-20 from the Filtros Company. Transmission coefficient is measured with two transducers while the reflection coefficient is measured with one transducer operating in a pulse-echo mode. In this work, the inverse problem is solved in the Bayesian framework which is well-suited to handle measurement and model uncertainties. All the unknown quantities are modeled as random variables. Furthermore, the prior models can be formulated so that they carry information of the target. Results of the characterization process agree with values found from the literature.

INVITED

Sound Intensity and Inverse Methods in Acoustics:

Paper ICA2016-486

Inverse methods of determining the acoustical parameters of porous sound absorbing metallic materials

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Abstract

Porous metallic material has become a kind of typical multifunctional material with the quick development of advanced manufacturing and forming process technologies in past years. At present, it has also been a structural function material with the characteristics of ultra-light weight, high specific strength and rigidity. As a sound absorbing material in noise control engineering, the acoustical parameters of porous metallic materials, especially such as the sinuosity factor, viscous and thermal characteristic lengths, are relatively difficult to obtain directly from the measurements under the common experimental conditions. On the contrary, the sound reflection and absorption coefficients, and the normal surface acoustic impedance of porous metallic materials are easy to exactly measure in a standard acoustic impedance tube. As a result, this paper introduced the inverse methods in order to determine the acoustical parameters of porous metallic materials based on available sound

absorbing models such as Johnson-Champoux-Allard model, Biot-Allard model, etc. Moreover, inverse methods based on Taboo search algorithm (TSA), simulated annealing genetic algorithm (SAGA) and linear regression (LR), were developed; by which the main acoustical parameters of some porous metallic materials were found out for use elsewhere as well. Finally, the results from the different inverse methods were in comparison with each other to validate the effectiveness of the aforementioned inverse algorithms.

Thursday morning, 8 September 2016 **Cardenal Pironio Auditorium**
09:00 - 10:40
Architectural Acoustics - Room and Building Acoustics
AA6 - Concert hall acoustics

INVITED
Concert Hall Acoustics:
Paper ICA2016-606

The acoustics of the concert hall Auditorio Juan Victoria from San Juan, Argentina

Ernesto Accolti[a,b],Yésica Alamino Naranjo[b], Alción Alonso Frank[b], Ernesto Kuchen[b]
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Abstract
The Auditorio Juan Victoria is a concert hall located in the homonym cultural building inaugurated in 1970, in San Juan province, Argentina. It seats 976 on an audience area of rectangular plant. The scenario is fan shaped and has the capacity for 80 seated musicians and 90 choristers standing. Hall dimensions are of about 22 m width, 40 m length and 10 m height. The hall is equipped with a pipe organ with 44 ranks and 3 565 pipes. In this article, the acoustic quality of the hall is assessed by a questionnaire. Measurements are taken using the state of the art methods, including ISO 3382-1 parameters. Results are compared with subjective and objective data from other similar halls and recommended values from literature.

Concert Hall Acoustics:
Paper ICA2016-166

Acoustic characterization of the Usina del Arte Symphony Hall

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Abstract
The old industrial building “Usina Don Pedro de Mendoza” was for 80 years one of the most important electric power stations of Buenos Aires city. In the late 90s the building was closed and abandoned until 10 years later, when it was redesigned as an arts center. This paper presents a historical and architectural review as well as an acoustic characterization of the new Usina del Arte Symphony Hall, the main concert hall of the city and the definitive host of the Symphonic and Philarmonic National orchestras. The equipment and protocols are carried out according to ISO 3382 in order to obtain multiple impulse responses, including monoaural, binaural and 3D sound field recordings. The acoustic parameters evaluated are RT30, EDT, D50, C80, STI, RaSTI, IACC and background noise (NC curve), including 3D impulse response mapping. The study also includes an objective evaluation of its sound field diffusivity using the SFDC (sound field diffusion coefficient).

Concert Hall Acoustics:

Paper ICA2016-729

Sound energy distribution in Italian opera houses

Massimo Garai[a], Simona De Cesaris[a], Federica Morandi[a], Dario D'Orazio[a]
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Abstract

A typical Italian opera houses is a complex system of coupled volumes: fly tower, orchestra pit, cavea (the volume where the stalls are), boxes, loggione (gallery). The way of propagation of the sound energy between one volume and the others is still a subject of research. The present work gives a contribution to the discussion by applying the Barron's revised theory to the analysis of recent measurements done in several Italian theatres. The averaged values of sound strength vs the distance from the sound source is plotted inserting in the Barron's equation either the classical reverberation time or the early decay time and different volume values. The significance of the different choices and their agreement with the experimental values are discussed. It is concluded that the spatial distribution of sound strength depends on the sound source position.

Concert Hall Acoustics:

Paper ICA2016-95

Acoustics for amplified music and a new, variable acoustics technology that includes low frequencies

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Abstract

Surveys among professional musicians and sound engineers reveal that a long reverberation time at low frequencies in halls during concerts of reinforced music such as pop and rock, is a common cause for an unacceptable sounding event. Mid- and high-frequency sound is seldom a reason for lack of clarity and definition due to a 6 times higher absorption by audience compared to low frequencies, and a higher directivity of speakers at these frequencies. Lower frequency sounds are, within the genre of popular music, rhythmically very active and loud, and a long reverberation leads to a situation where the various notes and sounds cannot be clearly distinguished. This reverberant bass sound rumble often partially masks even the direct higher pitched sounds. A new technology of inflated, thin plastic membranes presents a solution to this challenge of needed low-frequency control. It is equally suitable for multipurpose halls that need to adjust their acoustics by the push of a button and for halls and arenas that only occasionally present amplified music and need to be treated just for the event. This paper presents the authors' research as well as the technology showing applications in dissimilarly sized venues, including before and after measurements of reverberation time versus frequency.

Concert Hall Acoustics:

Paper ICA2016-664

Using a spherical microphone array for stage acoustics: A preliminary case for a new spatial parameter

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Abstract

The acoustic conditions on stage for musicians are traditionally assessed with an omnidirectional receiver; however, with the use of a spherical microphone array the directionality of on-stage sound fields can be examined. This paper explores the issues around using such a microphone for stage acoustic measurements. As part of this study the 32-channel spherical microphone array Eigenmike has been used for acoustic measurements on-stage in six Australian auditoria; additionally, in four of these venues a traditional omnidirectional receiver was also used. This paper compares the results of standard omnidirectional parameters with the Eigenmike and omnidirectional receiver to assess the

validity of the omnidirectional parameters derived from measurements with the Eigenmike. For example, the stage acoustic parameters the ‘support measures’ deviate between the microphones by no more than 0.5 dB. Additionally, the paper explores redefining the standard acoustic parameters to consider directionality, and presents these results in comparison to subjective musician assessments. A new parameter is proposed that corresponds well with the preferences of musician playing in ensemble. This work is being completed as part of larger study examining stage acoustics for chamber orchestras, which has also included subjective musician surveys with the Australian Chamber Orchestra regarding the venues included in the objective acoustic study.

Thursday morning, 8 September 2016 **Cardenal Pironio Auditorium**
11:00 - 12:00
Architectural Acoustics - Room and Building Acoustics
AA6 - Concert hall acoustics

Concert Hall Acoustics:
Paper ICA2016-732

Recordings of Italian Opera orchestra and soloists in a silent room

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Abstract
Anechoic recordings of symphony orchestra have been proposed in the literature and have been used in a multitude of studies concerning both innovative measurements and psychoacoustic experiments. Using the same approach, the present work shows the results of a recording campaign focused on the Italian Opera. Different motifs from Italian Operas have been played by professional musicians and soloist in the silent room of the Bologna University. The excerpts have been chosen both because of their musical style characteristics and their acoustic properties (dynamics, tymbre, vibrato). The chosen motifs come from scores of Donizetti, Verdi and Puccini, in order to consider various orchestrations and Opera styles.

Concert Hall Acoustics:
Paper ICA2016-32

Sound is a wave. A new concept of Huygens acoustic diffuser

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Abstract
We know that the reference rooms have diffraction to some extent which, has always believed to be good, because the diffuse energy delivered is weaker than a mirror sound energy produced above a smooth wall. So in our case we will explain the importance of the diffraction effects to produce more and better diffusion of sound. Here we will formulate the existing essential difference between the diffraction effect of Huygens and scattering effect of Schroeder. We will distinguish between Huygens diffraction plates versus scattering plates defined by M.R. Schroeder. In our defense of Huygens diffraction we have many cases where we have put to an experimental test, the principle of Christian Huygens. After of these experiences we can conclude: Sound is a wave and is not a sound ray, and this has become clear in our various acoustic experiments conducted by us. We have proved that the solution of the acoustics of many halls analyzed by us, had been treated thinking that the sound are waves, but not rays. Therefore our opinion is, that in future must be required solve many acoustic problems in halls using mathematical and geometrical methods treating with sound waves. Because it, with the raytracing method is not possible to solve these problems; nor with using the scattering factors, due to this system only is valid in a limited frequency range.

Concert Hall Acoustics:

Paper ICA2016-754

Analysis of lightweight acoustic reflectors

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Abstract

Halls for music performance frequently require the design of acoustic reflectors to redirect the sound waves toward the audience. In order to get adequate reflecting properties the usual criterion is that the surface density be no less than 20 kg/m^2. However, this may prove too heavy in certain cases, so other possible solutions must be studied. In this paper, light-weight reflectors, such as medium density fibreboard or plywood panels clamped at their boundaries, where structural rigidity replaces mass at the low frequency end, have been investigated. A compliance model is compared with the mass model showing that the structure is suitable.

Thursday morning, 8 September 2016 **Room 204**
09:00 - 10:40
Communication Acoustics
CA1 - The Technology of Binaural Listening and Understanding

INVITED

The Technology of Binaural Listening & Understanding:

Paper ICA2016-363

Auditory illusion through headphones: History, challenges and new solutions

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Abstract

The dream of perfect recreation of sound has always consisted of two parts: Reproduction of monaural sounds such that they seem to be exact copies of an original signal and the plausible recreation of complex sound environments, the possibility to be immersed in sound. The latter goal seems to be much more difficult, especially if we consider reproduction over headphones. From standard two-channel sounds reproduced over headphones through artificial head recordings, the inclusion of HRTF and binaural room impulse responses, always something was missing to create a perfect auditory illusion. Depending on refinements like individually adapted HRTF etc. these methods work for many people, but not for everybody. As we know now, in addition to the static, source and listener dependent modifications to headphone sound we need to pay attention to cognitive effects: The perceived presence of an acoustical room rendering changes depending on our expectations. Prominent context effects are for example acoustic divergence between the listening room and the synthesized scene, visibility of the listening room, and prior knowledge triggered by where we have been before. Furthermore, cognitive effects are mostly time variant which includes anticipation and assimilation processes caused by training and adaptation. We present experiments proving some of these well-known contextual effects by investigating features like distance perception, externalization, and localization. These features are shifted by adaptation and training. Furthermore, we present some proposals how to get to a next level of fidelity in headphone listening. This includes the use of room simulation software and the adaptation of its auralization to different listening rooms by changing acoustical parameters.

INVITED

The Technology of Binaural Listening & Understanding:

Paper ICA2016-715

Multimodal fusion and inference using binaural audition and vision

Benjamin Cohen-Lhyver[a;b], **Sylvain Argentieri**[a;b], **Bruno Gas**[a;b]

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Abstract

Hearing is a key modality on which several perceptual human processes rely on. Together with vision, these two modalities offer a 360 degrees wide, highly sensitive, quickly adaptive, and incredibly precise system of perception of the environment. In an exploratory robotics context, the concept of audiovisual objects is very relevant for a robot since it enables it to better understand its environment, and also to interact with it. However, how to face the cases when an object is out of sight, or when it does not emits sound, that is, the cases of missing information? The proposed Multimodal Fusion and Inference (MFI) system takes advantages of having (i) multimodal information and (ii) the ability to move in the environment, to implement a low-level attentional algorithm that enables a mobile robot to understand its environment in terms of audiovisual objects. In the case of a missing modality, the proposed algorithm is able to infer the missing data thus providing to the robot full information to higher cognitive stages. The MFI system is based on an online and unsupervised learning algorithm using a modified self-organizing map. Furthermore, the MFI exploits the ability to turn the robot head towards objects, thus benefiting from active perception to reinforce autonomously what the system is actually learning. Results exhibits promising performances in closed-loop scenarios involving sound and image classifiers.

INVITED

The Technology of Binaural Listening & Understanding:

Paper ICA2016-827

A precedence effect model with top-down processing stages based on visual cues

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Abstract

An audiovisual model is introduced to demonstrate the potential benefit of having visual knowledge about room dimensions when localizing a sound source in the presence of early reflections and diffuse reverberation. The model uses a basic perceptual model of computer vision to detect room edges from a stereoscopic input signal. The visual part of the model can also be used to predict the expected reverberation time and direct-to-reverberant signal-amplitude ratio based on psychophysical data. The auditory precedence effect model uses a dual-layer cross-correlation/auto-correlation algorithm to determine the localization cues of the direct sound source and estimates a binaural activity pattern for both the direct sound source and early reflections. The visual part of the model estimates angles of incidence and delays for the first two side reflections of a given frontal sound source. It then determines the azimuth angles and distances from the corners of a stereoscopic visualization of the room to determine the arrival times of two early side reflections to optimize the integration window for the cross-correlation process. It is demonstrated that the model benefits substantially, reducing the average azimuth error by up to 30º.

INVITED

The Technology of Binaural Listening & Understanding:

Paper ICA2016-718

Contributions of binaural processing to segregating and selecting speech in a complex sound mixture

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Abstract

Intuitively, we all believe that binaural processing plays a critical role in communication, especially at the venerable "cocktail party." Indeed, if you attend a poster session at a large conference (like the ICA), close your eyes, plug one ear, and try to follow a scientific discussion, you will experience the importance of having two ears. Here we will discuss how binaural processing contributes to two key aspects of understanding speech in crowded settings: focusing attention on whichever source is important in the sound mixture (selection), and separating that source from other sources in the mixture (segregation). Behavioral data show that binaural cues help with source selection, or focusing of auditory attention. When it comes to sound segregation, the contributions of binaural hearing depend on the time scale that one considers. For segregating one speech syllable from a sound mixture, the data suggest that binaural cues are not very salient; they are overridden by other cues such as common onset and offset. However, when connecting together syllables into a continuous stream of speech, spatial cues play a much stronger role. Understanding the role of binaural hearing, and the time scales on which spatial cues matter, perceptually, can guide how binaural cues are used in hearing devices, such as hearing aids, to improve speech understanding in everyday settings.

INVITED

The Technology of Binaural Listening & Understanding:

Paper ICA2016-633

Binaural technology and automatic speech recognition

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Abstract

It is well known that binaural processing is very useful for separating incoming sound sources as well as for improving the intelligibility of speech in reverberant environments. This paper will describe and compare a number of ways in which automatic speech recognition accuracy in difficult acoustical environments can be improved through the use of signal processing techniques that are motivated by our understanding of binaural perception and binaural technology. These approaches have been inspired by the classic models of interaural cross-correlation proposed by Jeffress and elaborated on by many others, which have been applied to describe many binaural phenomena. They are also motivated in part by the precedence effect, in which the earliestarriving components of a complex signal dominate perception. We compare the performance of a number of methods that use two or more microphones to improve the accuracy of automatic speech recognition systems operating in cluttered, noisy, and reverberant environments. Typical implementations differ in the extent to which practical engineering solutions adhere to classical binaural modeling, in the specific processing mechanisms that are used to impose suppression motivated by the precedence effect, and in the precise mechanism used to extract interaural time differences. We demonstrate that the use of binaural-based processing can provide substantially improved speech recognition accuracy in noisy, cluttered, and reverberant environments compared to baseline delay-and-sum beamforming. The type of signal manipulation that is most effective for improving performance in reverberation is different from what is most effective for ameliorating the effects of degradation caused by spatially-separated interfering sound sources.

INVITED

The Technology of Binaural Listening & Understanding:

[Paper ICA2016-299]

Spatial modulation: Hearing the environment

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Abstract

Auditory processing of complex sources, after an initial peripheral spectro-temporal stage, is thought to have a more central stage identify in the output time segments and frequency regions of higher activity by way of a temporal and spectral modulation analysis. Such analysis broadens the view on perception, both that of complex signals and of auditory scene analysis (ASA). When resolution of temporal and spectral modulations is adequate, the auditory system can decode complex signals and separate simultaneous sources in a scene. Although research in the modulation domain has uncovered important properties of the central (cortical) mechanism active in such analysis, so far it has bypassed the spatial dimension. The present study proposes to include spatial modulation in the horizontal plane into this mechanism. The signal emanating from multiple and diverse sources at different azimuths will first undergo peripheral binaural processing using known methods, consisting of frequency analysis, phase-compensated rectification, left-right cross-correlation, straightening, and weighted frequency integration. The output will represent azimuthal activity between $-\pi$ and $+\pi$ radians as a function of time. This analysis stage will be followed by the modulation analysis stage: convolution of the magnitudes, across the azimuth activity axis, with a kernel function that signifies resolution of nearby simultaneous sources. Results of the spatial modulation analysis will be shown as a function of the same input frequency analyzed and put through a stage of temporal modulation processing. Spatial and temporal modulation analysis results viewed side-by side will predict the temporal fluctuation rate and spatial source density at which perception of multiple sources should be optimal.

INVITED

The Technology of Binaural Listening & Understanding:

[Paper ICA2016-154]

Simulating cognitive feedback in the context of binaural scene analysis

Thomas Walther[a], Jens Blauert[b]

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Abstract

Dynamic auditory scene analysis (DASA) requires decisions on a cognitive level, for example, when assigning meaning to scene elements and/or interpreting scenes to induce appropriate actions. To this end, feedback from the cognitive level to the auditory-signal processing level has to be considered. In our talk, a software system is described to be used to develop and test actions of robots in search-and-rescue (SAR) scenarios. In these scenarios a victim in a (moderately) complex environment has to be identified and localized. The actions are predominantly based on binaural cues, derived from the ear signals of a head-and-torso simulator (dummy head) that can actively move about in the scene to be explored. Data that cannot yet be acquired from the auralized scenario at the current state of development of our system, are emulated to enable demonstration of system functionality. Further, visual cues may be employed for assistance if necessary, since the virtual dummy head is equipped with a stereo camera. In summary, our software system, which has been set up in the context of the European-Union FET project TWO!EARS (ICT-618075, twoears.eu), provides a virtual environment for testing the routines necessary to accomplish the tasks mentioned above.

INVITED

The Technology of Binaural Listening & Understanding:

Paper ICA2016-346

Assessment of audio quality and experience using binaural-hearing models

Alexander Raake[(a)], **Hagen Wierstorf**[(a)]

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Abstract

The paper presents results on spatial audio quality evaluation and approaches for modeling the data using binaural-hearing models, following the interactive and object-related framework developed in the EC-funded FET-Open project TWO!EARS (www.twoears.eu, see also Raake & Blauert QoMEX 2013, Raake et al. Forum Acusticum 2014). Two types of tests and respective modeling approaches are presented: (1) Feature modeling following the well-established approach of spatial and timbral fidelity prediction. To this aim, a feature-specific set of listening tests was conducted for ground-truth data collection. The resulting feature models apply different parts of the bottom-up auditory processing modules from the TWO!EARS framework. (2) Preference modeling. For collecting the underlying data, a listening test series was conducted applying tailored mixes for different musical pieces with different variations of the mixing choices for specific sources in the scene. Using a full paired-comparison test paradigm, preference ratings were collected from listeners. The respective model involves TWO!EARS' scene segregation for object identification and subsequent object-specific feature extraction.

Thursday morning, 8 September 2016 **Microcinema**

09:00 - 10:40

11:00 - 12:00

Course

ACOUSTIC DESIGN OF MUFFLERS

Lecturer

Dr. Tamer Elnady

Associate Professor, Group for Advanced Research in Dynamic Systems (ASU-GARDS)

ASU Sound & Vibration Lab.

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URL: www.asugards.edu.eg

Brief Summary

Most muffler design is accomplished by using “cut-and-try” methods that rely on what has worked in the past and/or extensive full-scale testing on engines for validation. This approach can be extremely costly, in particular with regard to large industrial mufflers, in terms of testing or in other losses caused by inability of the design to meeting the project criteria. New computer software aimed at muffler design can shorten the design cycle and yield more effective results. This seminar provides an introduction to the behavior of mufflers and silencers including a description of the two-port approach to muffler design. The seminar covers the acoustic simulation of muffler and silencer systems and the use of experimental methods to measure muffler performance. Following a review of basic muffler concepts and definitions, the seminar will focus on meeting design objectives such as insertion loss with a specified back pressure requirement. It will show how modern software such as SIDLAB can be used to model both the acoustics and flow in achieving the design objective and the role that 1D engine simulations can play in providing important input. The final topic will cover optimization of muffler design to meet a specified design objective with a specified space constraint. The main focus is on large and small IC-engine intake and exhaust systems, but most of the information is also applicable to any pipe or duct system.

Psychological and Physiological Acoustics (others):
Paper ICA2016-885

More robust estimates for DPOAE level at audiometric frequencies

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Abstract
Current clinical methods determine 2 f1 f2 distortion product oto-acoustic emission (DPAOE) levels at discrete frequencies, and often only at the audiometric standard frequencies in order to save time. The measured result is known to be a superposition of at least two components, the generator component originating from a region around the primary f2, and the reflection component from the 2 f1 f2 site. Distinct interference patterns in high resolution DPOAE data reveal that these two components can be of similar magnitude, and periodically cancel each other entirely. When measurements are made at only few frequencies, there is a risk to find one or more low amplitude measurement, even in a healthy ear with otherwise high emissions. In the present study, data from previous studies measured with a high frequency resolution is used for simulating a better use of measurements at and around the audiometric frequency. A ”local” model of the two component superposition is applied, and the trade-off between measurement time, and robustness of the measure is discussed.

Psychological and Physiological Acoustics (others):
Paper ICA2016-219

Investigation of hearing perception at ultrasound frequencies by functional magnetic resonance imaging (fMRI) and magnetoencephalography (MEG)

Robert Kühler[a], Markus Weichenberger[b], Martin Bauer[a], Simone Kühn[b], Tilmann Sander-Thömmes[a], Albrecht Ihlenfeld[a], Bernd Ittermann[a], Johannes Hensel[a], Christian Koch[a]
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Abstract
Airborne ultrasound is applied in many technical and medical processes and has increasingly moved into daily life. Because of a potential exposure of humans the question whether sound at these frequencies can be heard and whether these sounds can be of any risk for the hearing system or for wellbeing and health of an individual in general, is of great practical relevance. To study these issues audiological methods and neuroimaging were combined in order to obtain an objective rationale of the auditory perception of airborne ultrasound in humans. In a first step the monaural pure-tone hearing threshold for 26 young test subjects (19 – 33 years) in the frequency range from 14 to 24 kHz was determined. The hearing threshold values rose steeply with increasing frequency up to around 21 kHz followed by a range with smaller slope towards 24 kHz. In a next step neuroimaging techniques were applied to find brain activation following the stimulation by ultrasound between 20 and 24 kHz. Functional magnetic resonance imaging (fMRI) with sound pressure levels slightly above and below individual threshold was used in experiments with the same test persons as in the audiological measurements. Although test subjects reported audible sensation no brain activation could be identified in the above-threshold case except for the lowest test frequency at 14 kHz. Magnetoencephalography (MEG) was employed as an alternative method with the same test person group. Brain activation was measured, but again no auditory cortex activation was found above 14 kHz.

Psychological and Physiological Acoustics (others):

Paper ICA2016-832

Hybrid method for obtaining individual head related transfer functions (HRTF): pinna molding and head-torso photogrammetric 3D reconstruction

Sebastián Fingerhuth[a], Juan Barraza[a], Danny Angles[a]
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Abstract

In this work we present the results of a method of creating individualized 3D CAD models of heads and *pinna*. It is mainly an analysis of the quality of the results. Photogrammetry is used to obtain a 3D CAD model of the head of a subject from a set of photographs taken from three different orbits. To obtain the model of the *pinna*, first a mold of it has to be created, which than is photographed in a similar way as the head.For the analysis different settings, parameters and configurations of the set-up and of the software were tested and compared. Also different photo sessions for the same subject were done and the final results compared, to test the repeatability and robustness of the method. Distances measured on the face of the subject as well as on the CAD model were compared and a mean error value was computed. The results show that the mean error is below 5 %.

Thursday morning, 8 September 2016 **Auditorium 3**

09:00 - 10:40

11:00 - 12:00

Course

ULTRASOUND, CAVITATION, SONOCHEMISTRY

Lecturer

Sivakumar Manickam, PhD Tech., FHEA (UK)

Professor of Chemical and Nanopharmaceutical Process Engineering Head, Manufacturing and Industrial Processes Research Division Director, Centre for Nanotechnology and Advanced Materials (CENTAM)

Department of Chemical and Environmental Engineering Faculty of Engineering, University of Nottingham Malaysia Campus

Brief Summary

In these days green and clean processing techniques receive greater attention in a wide variety of technologies. In this connection, this short course offers an excellent opportunity and intends to expose the participants to learn about the fundamentals, effective usage and technological applications of ultrasound. With proper learning and understanding, ultrasound could be employed in an energy efficient way. Besides, the discussion will be made about the translation of lab scale to plant scale. The presenter has a very long experience in designing ultrasound reactors and thus it is an opportunity to learn this technique in an easy way through this short course.

Thursday morning, 8 September 2016 Lounge Lateral Room
10:00 - 10:20
POSTER SESSION - Monitor 1
Musical Acoustics
MU2 - String Instruments

POSTER
String Instruments:
Paper ICA2016-821

Improved frequency-dependent damping for time domain modelling of linear string vibration

Charlotte Desvages[a], Stefan Bilbao[b], Michele Ducceschi[c]
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Abstract
Lossy linear stiff string vibration plays an important role in musical acoustics. Experimental studies have demonstrated the complex dependence of decay time with frequency, confirmed by detailed modelling of dissipated power in linear strings. Losses at a particular frequency can be expressed as a function of the physical parameters defining the system; damping due to air viscosity is predominant at low frequencies, whereas internal friction prevails in the higher frequency range. Such a frequency domain characterisation is clearly well-suited to simulation methods based on, e.g., modal decompositions, for experimental comparison or sound synthesis. However, more general string models might include features difficult to realise with such models, in particular nonlinear effects. In this case, it is useful to approach modelling directly in the space-time domain. This work is concerned with the translation of the frequency domain damping characteristics to a space-time domain framework, leading, ultimately, to a coupled system of partial differential equations. Such a system can be used as a starting point for a time-stepping algorithm; an important constraint to ensure numerical stability is then that of passivity, or dissipativity. Candidate loss terms are characterised in terms of positive real functions, as a starting point for optimisation procedures. Simulation results are presented for a variety of linear strings.

Thursday morning, 8 September 2016 Lounge Lateral Room
10:20 - 10:40
POSTER SESSION - Monitor 1
Musical Acoustics
MU3 - Numerical Computation in Musical Acoustics

POSTER
Numerical Computation in Musical Acoustics:
Paper ICA2016-576

Transmission-line matrix modeling and transformed area functions of flaring musical horns

Felipe Orduña-Bustamante[a], Pablo Luis Rendón, Enedina Martínez-Montejo
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Abstract
Practical application to the numerical analysis of musical horns is presented in this work of the well known transmission-line matrices for the Salmon family of flaring horns, which include cylindrical, conical, exponential, hyperbolic and trigonometric profiles. Geometrical formulations are presented of the progressive spherical wavefront approximation, which has been shown to be a necessary step for the sufficiently accurate application of Webster's wave equation to rapidly flaring horns. This leads to a

necessary transformation of the horn area function, from the usual flat cross-sectional area in terms of the axial coordinate, into a curved cap-like wavefront area as a function of the arc-length coordinate along the horn profile. Results are presented of the input acoustical impedance of a trumpet and a trombone, for which numerical and experimental results are compared.

Thursday morning, 8 September 2016 **Lounge Lateral Room**
11:00 - 11:40
POSTER SESSION - Monitor 1
Psychological and Physiological Acoustics
PP1 - Free-Field Virtual Psychoacoustics and Hearing Impairment

POSTER

Free-Field Virtual Psychoacoustics and Hearing Impairment:

Paper ICA2016-53

Measurement of pinna flare angle and its effect on individualized head-related transfer functions

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Abstract

Head-related transfer functions (HRTFs) are essential to the researches of binaural hearing and applications of virtual auditory display. Generally, HRTFs vary with frequency as well as source position relative to head centre. They also depend on anatomical structure and parameters of individual subject. The related anatomical parameters mainly include the dimensions of head and pinnae, and the position parameters between the pinna and head. In present work, the influence of pinna flare angle on HRTFs is investigated by using numerical calculation. Based on a combination model of ellipsoidal head and laser-scanned pinnae from KEMAR artificial head, the pinna flare angle in the model is changed from an original value with displacements of ±6.7 degrees, and the corresponding horizontal HRTFs at various azimuths are calculated by using boundary element method. Results indicate that the changing pinna flare angle leads to a systemic variation on the frequency and azimuthal distribution of HRTF magnitude spectra. And approximately, the variation of azimuthal distribution of HRTF magnitude spectra is linearly related to the change of pinna flare angle. Therefore, individualized azimuthal HRTF magnitudes corresponding to various pinna flare angle can be predicted or customized by applying appropriate azimuthal rotation manipulation to a set of original HRTF data with certain pinna flare angle. The method in present work is applicable to customize a matched set of HRTFs for improving the perceived performance of virtual auditory display.

INVITED

POSTER

Free-Field Virtual Psychoacoustics and Hearing Impairment:

Paper ICA2016-816

Are the precedence effect and spatial impression the result of different auditory processes?

M. Torben Pastore[a], Elizabeth Teret[c], Brandon Cudequest[c], Jonas Braasch[c]
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Abstract

The sensation of auditory spaciousness is closely related to the pattern of reflections in a room, often described by a room impulse response (RIR). The early and late portions of RIRs are correspondingly the basis of most metrics of auditory spatial impression. Even though this pattern of reflections might reasonably

be expected to indicate many sound source locations, listeners commonly localize sounds to their sources. This phenomenon, called the precedence effect (PE), is often thought to result from the suppression of reflections. This appears to present a paradox. How do we gain spatial information from reflections we are thought to suppress? Also, is this process different for early and late reverberation? In three parallel studies, we addressed this question by examining the specific roles of specular (early) and diffuse (usually late) reflections. The first experiment compared listener performance under conditions that elicited the precedence effect with diffuse or specular reflections. The second investigated listeners' ability to match reverb times using different types of stimuli. The third compared apparent source width (ASW) resulting from physically wide sources to narrow sound sources with side reflections. Our results suggest that the PE, ASW, and listener envelopment are the result of closely related auditory processes. We also find that listeners do not appear to have a clear internal temporal representation of the decaying late reverb tail of a room impulse response. Possible implications for current metrics of spatial impression are considered.

Thursday morning, 8 September 2016 **Lounge Lateral Room**
11:40 - 12:00
POSTER SESSION - Monitor 1
Psychological and Physiological Acoustics
PP2 - Product Sound Quality and Multimodal Interaction

POSTER

Product Sound Quality and Multimodal Interaction:

Paper ICA2016-851

Sound quality approach for detecting abnormal noise of dehumidifier

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Abstract

Sound Quality becomes a new trend in the field of product sound. Existing noise control focuses to decrease or remove radiated noise level from a product. However, in these days, the objective of noise control is to optimize product sound in the viewpoint of acceptability on its function and subjective feeling. For this reason, sometimes, sound quality turns into a criterion to express product quality. The purpose of this study is to develop sound quality index or factor for detecting abnormality of the noise of dehumidifier. To this end, the noises from 19 normal and 6 abnormal dehumidifiers were recorded and the operation condition was classified into three groups. Among the three abnormal sounds: motor, rattle, refrigerant, motor noise was dealt with as target abnormal sound. The degree of abnormality of the sounds was evaluated through the subjective listening test. In the objective manner, some SQ metrics like loudness, sharpness, and roughness were calculated and masking effect between peak components was also considered. As a result, it was investigated that excessive motor noise was dependent on the strength of peak components related to the fan motor operation, and then could be detected by SPL excess.

Thursday morning, 8 September 2016 Lounge Lateral Room
10:20 - 10:40
POSTER SESSION - Monitor 2
Virtual Acoustics
VA1 - Virtual Acoustics

POSTER
Virtual Acoustics:
Paper ICA2016-342

An asynchronous HRTF measurement method based on phase alignment

Mengfan Zhang[(a)], Fengyun Zhu[(a)], Tianshu Qu[(a)], Xihong Wu[(a)]
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Abstract
The virtual sound technology based on the Head-Related Transfer Function (HRTF) is important in many applications such as gaming, education and military. Currently, the most important and accurate method to obtain HRTF is experimental measurement. In experimental measurement, the Maxim Length Sequence (MLS), the sweep signal, and impulse signal are usually used as the exciting signal and played in loop to generate several HRTFs and then the HRTFs are averaged to improve the Signal to Noise Ratio (SNR) of the results. However, the inconsistency of the timing modules (Oscillator) used in the recording system and in the playback system results the time non-alignment between the measured HRTFs in different played loop. The time non-alignment destroys the function of the average process and makes the average results distortion. For solving this problem, this paper proposed an asynchronous HRTF measurement method based on the phase alignment. Firstly, HRTFs are measured using the MLS signal as the exciting signal which are played in loops; then the phase alignment algorithm is applied to HRTFs in each loop to compensate the difference of the timing modules (Oscillator) of the recording system and the playback system; lastly, the aligned HRTFs are averaged to generate the result HRTF. The evaluation experiment results show that the Peak-SNR of the proposed HRTF measured method are improved about 4.5 dB compare to that of the traditional method.

Thursday morning, 8 September 2016 Lounge Lateral Room
11:00 - 12:00
POSTER SESSION - Monitor 2
Signal Processing in Acoustics
SP4 - Signal Processing in Acoustics (others)

POSTER
Signal Processing Acoustics (others):
Paper ICA2016-125

Inter-channel transfer function based parametric stereo coding system

Qingbo Huang, Tianshu Qu, Liang Li, Xihong Wu
Key Laboratory on Machine Perception (Ministry of Education), Speech and Hearing Research Center, PekingUniversity, Beijing, China, qutianshu@pku.edu.cn

Abstract
Traditionally, most parametric stereo coding systems use the inter-channel level difference (ILD) and the inter-channel correlation (ICC) as the side information to compress the stereo signal effectively. In this paper, a novel parametric stereo coding method is proposed by using the inter-channel transfer function (ITF) as the side information. The ITF is defined as the transfer function from the mixed signal (the sum of the left channel signal and the right channel signal) to the difference of the left and right channel signal. The ITF contains more spatial information of the stereo sound compared to the ILD and the ICC do. The ITFs of all the frames are grouped together to construct the two dimension matrix. Then, the discrete cosine transform is used to compress the ITFs matrix according to the

desired bitrates. Therefore, the redundancies of the stereo signals are diminished not only in the frequency domain but also in the time domain. Lastly, the subjective evaluation experiments based on MUSHRA were carried out to compare the proposed system with the HE-AAC system. The results showed that the proposed system performed comparably to HE-AAC system in the speech signals, the transient musical signals, and the steady state musical signal.

POSTER

Signal Processing in Acoustics (others):

Paper ICA2016-303

A control of maximum demodulation distance with gas layer for parametric loudspeaker

Kirara Ariyoshi[(a)], Shinya Komori[(b)], Takahiro Fukumori[(c)], Masato Nakayama[(d)], Takanobu Nishiura[(e)]

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Abstract

A parametric loudspeaker can transmit an acoustic wave to only a particular listener by utilizing an ultrasound wave. The parametric loudspeaker utilizes the amplitude modulated (AM) wave which is designed by modulating amplitude of the ultrasound wave with an audible sound wave. The intense AM wave gradually demodulates into the original audible sound wave by the nonlinear interaction in the air. The sound pressure level (SPL) of the demodulated audible sound wave depends on the demodulation rate of the intense AM wave. The demodulation rate is defined as a ratio of the distance of maximum SPL to that between the parametric loudspeaker and the observed position. It is therefore difficult to give the demodulated sound wave with maximum SPL to the listener being at any position. The maximum demodulation distance depends on the density of a gas layer in which the AM wave is transmitted. Based on this principle, we propose a new method which controls the maximum demodulation distance by transmitting the AM wave through the gas layer whose density is different from the air. By using this method, it is possible to transmit the audible sound with maximum SPL to the listener at the target position. As a result of the evaluation experiment, we confirmed the effectiveness of the proposed method.

POSTER

Signal Processing in Acoustics (others):

Paper ICA2016-304

Distant-talking speech acquisition using optical measurement system based on speckle intensity

Tomoyuki Mizuno[(a)], Yukoh Wakabayashi[(b)], Takahiro Fukumori[(c)], Masato Nakayama(d)**, Takanobu Nishiura**(e)

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Abstract

Various microphones such as the parabolic and shotgun microphones have been developed for measuring the distant-talking speech. It is very important for security and criminal situation de- tection to capture the speech. However, it is difficult for general microphones to capture the distant-talking speech if there is an object between a speaker and a microphone. In this study, we focus that a thin object is slightly vibrated by the speech and try to obtain the speech signal by measuring the vibration of the object. We also propose an optical measurement system by converting the measured speckle intensity of the laser light to the electrical signal. In general, the speech signal is obtained with general microphones by converting a vibration of microphone diaphragm to an electrical signal. On the other hand, the proposed system obtains the electrical signal by utilizing the thin object as an external diaphragm. Therefore, the electrical signal is distorted depending on a material of the vibrated object. In addition, the signal is also degraded by internal noise which arises in the circuit and external noise

which is the affection of the un- wanted vibration. The proposed system thus applies the digital signal processing for reducing the distortion. Finally, we confirmed the effectiveness of the proposed system through evaluation experiments.

Thursday midday, 8 September 2016 **Juan Pablo II Auditorium**
12:00 - 13:00
Plenary Lecture:
Chair: Luís Godinho

Barbara Shinn-Cunningham

Paper ICA2016-915

How the brain makes sense of complex auditory scenes

Barbara Shinn-Cunningham
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Abstract
Everyday listening involves a complex interplay between the ear, which transduces sound energy into neural responses, and the brain, which makes sense of these inputs. Historically, research on the ear tended to ignore the fact that what we can perceive in sound depends on what task the brain is engaged by, while research on cortical processing of sound ignored the complexity and sophistication of how the ear works. In this talk, I will explore how everyday perceptual abilities depend jointly on how the ear encodes information (and individual differences in the fidelity with which it does so) and how attention and other state dependent variables change the information we perceive.

Thursday afternoon, 8 September 2016 **Juan Pablo II Auditorium**
14:30 - 15:30
Noise: Sources and Control
NS2 - Hearing Protectors

Hearing Protectors:
Paper ICA2016-98

Using finite–element modeling to predict the effect of sound incidence on the Noise Reduction based attenuation of earmuffs

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Abstract
The sound attenuation of Hearing Protector Devices (HPD) can be measured objectively using the “Microphone In the Real Ear” (MIRE) method or with its field counterpart, the F-MIRE method. This two microphone method, with one microphone outside and the other inside the HPD, can be used to evaluate the attenuation in the form of a noise reduction (NR) obtained from the difference between the sound pressure levels at the two microphones. Correction factors, based usually on diffuse field or frontal incidence, can be applied on NR values to obtain an estimate of the insertion loss (IL) of the HPD, a more common measure of the sound attenuation. The F-MIRE method has been used for continuous measurements on earmuffs in real field environments where the sound field can depart

significantly from a diffuse field and may show strong directionalities. These field attenuation measurements have been shown to depend on the direction of the incoming sound, therefore, asking for a revision of the correction factors. Obtaining experimentally these factors that depend on both the external and internal microphone positions together with the incidence angle is cumbersome. This paper uses a 3D finite element model of an Acoustic Test Fixture (ATF) with and without earmuff, excited acoustically at various angles of incidence to calculate the correction factors. To assess the validity of the model, the simulated attenuation results are compared with experimental measurements on an ATF equipped with an earmuff placed in an anechoic room for various incidence angles. The effect on the correction factors of the angle of incidence together with the positions of the external and internal microphones is then simulated and discussed.

Hearing Protectors:

Paper ICA2016-808

Measurement of insertion loss of ear-muff type protectors using the impulse response technique

Gabriel A. Cravero[(a)], Lucas G. Gilberto[(a)], Sebastián P. Ferreyra[(a)], Marina G. Cortellini[(a)], Fernando M. González[(a)], Mario R. Serra[(a)(b)]

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[(b)] Consejo Nacional de Investigaciones Científicas y Técnicas (CONICET), Buenos Aires, Argentina

Abstract

The essential requirements that hearing protectors need to fulfil as personal protective equipment are evaluated in Argentina through a series of IRAM (Instituto Argentino de Normalización y Certificación) standards. In particular, IRAM 4060-3 (equivalent to ISO 4869-3:2007) standard specifies a simplified method for the measurement of insertion loss (IL) of ear-muff type protectors. This method can be used to test occasional performance differences on manufacturing, as part of type approval or certification procedures, and to investigate performance variations due to product aging. The IL of a hearing protector is commonly obtained using systems based on broadband pink noise (BPN) signals. This paper analyzes and compares the IL of eight different hearing protectors obtained with this technique and with an impulse response measurement based system. The results show that both techniques present similar IL values, within the dispersion range expected for this test. Also, the impulse response system reduces the test execution time, and proved to be robust, reliable and economical.

Hearing Protectors:

Paper ICA2016-354

Uncertainty of hearing protector noise attenuation based on REAT method

Rafael Gerges[(a)], Samir Gerges[(b)], E. Felipe Vergara[(c)]

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Abstract

The accuracy of quality of the measurement is characterized by measurement uncertainty, which defines an interval around the measured value where the true value lies with some probability. Uncertainty estimates obtained as standard deviations of repeated measurement results are called A type uncertainty estimates. If uncertainty is estimated using some means other than statistical treatment of repeated measurement results then the obtained estimates are called B type uncertainty estimates. This is described in detail in the ISO GUM “Guide to express of uncertainty in measurements”. This paper presents the uncertainty of measurement hearing protector noise attenuation. The international standard technique for these measurements is based on measuring human hearing thresholds with and without using of hearing protector and is known as REAT method. Discussion is presented on the variation of

uncertainty which related to different type of sources, but mainly due to human subject response which is very strongly related to the fitting of the hearing protector.

Thursday afternoon, 8 September 2016 **Juan Pablo II Auditorium**
15:30 - 16:10
Noise: Sources and Control
NS3 - Launch Vehicle Acoustics

Launch Vehicle Acoustics:
Paper ICA2016-495

Reducing rear axle gear whine noise inside a car by influencing the structure-borne sound transfer path using structurally integrated piezo-actuators

Jan Troge[a], Welf-Guntram Drossel[a], Marco Lochmahr[b], Sebastian Zumach[a]
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Abstract
The gear whine noise of rear axles is a well-known acoustic phenomenon especially for rear-wheel and four-wheel drive vehicles. The acoustical optimization of this problem leads to the major goal conflict: improvement of the vibrational isolation of the rear axle versus excellent driving dynamics of the vehicle. A possible solution for this issue is in focus of a research project at Fraunhofer Institute for Machine Tools and Forming Technology IWU in cooperation with Mercedes AMG GmbH. The aim is to reduce the noise contributions of the rear axle inside the car using an active vibration control system on the transfer path from the axle into the vehicle based on structurally integrated piezo-actuators. This paper describes the development process of the active vibration control system. At first, a FEM-simulation model has been created which is able to represent the operational deflection shapes of the rear axle assembly in critical operating points. In a next step, two simulation approaches have been applied in order to identify promising excitation points for an actuator application: the structural intensity analysis and a sensitivity analysis of transfer functions of the rear axle structure. Furthermore, the geometry and material properties of the piezo-actuator have been implemented in the simulation to calculate the force reduction on the coupling points to the vehicle body. In addition, a rear axle test bench has been set up in order to reproduce the operational deflection shapes and validate the simulation results.

Launch Vehicle Acoustics:
Paper ICA2016-684

Projection algorithms for target spectrum matrix definition in MIMO direct field acoustic control

Mariano Alvarez Blanco[a], Karl Janssens[a], Fabio Bianciardi[a]
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Abstract
Reducing the cost and risk of spacecraft acoustic qualification tests make many of the space contractors prefer to conduct Direct-Field-Acoustic-eXcitation (DFAX) tests instead of Reverberant-Field-Acoustic-Tests (RFAT). The first consists in setting a loudspeaker array around the test specimen. DFAX test does not demand the construction of a large reverberant test facility dedicated exclusively for acoustic qualification as RFAT. However, reliably reproducing the characteristic diffuse field and the uniform sound pressure spatial distribution of RFAT still represents a challenge for DFAX. Close loop control methodologies were proposed to achieve this goal. Multiple-Input-Multiple-Output (MIMO) strategies seem to give more flexibility than Single-Input-Single-Output (SISO) to handle the acoustic wave interferences. Hence, they have enhanced the diffusivity and uniformity of the acoustic field for DFAX. However, MIMO demands a more complex test configuration, e.g. the test requirement must be defined as a target spectral density matrix (SDM) instead of a single power spectrum density (PSD) profile. Besides, a centralized MIMO controller is computationally more expensive than SISO. Regarding the target SDM definition, a recent publication has introduced the energy-sink phenomenon. It was shown, theoretically and

experimentally, how different target definitions lead to different control performances. In this research new target definition approaches are described. They are based on the projection of the test requirement into the subspace of achievable pressure responses. Experimental results show how the projection approaches improve the control performance of rectangular systems with MIMO control strategies.

Thursday afternoon, 8 September 2016 **Juan Pablo II Auditorium**
16:30 - 18:30
Noise: Sources and Control
NS4 - Materials for Noise Control

Materials for Noise Control:
Paper ICA2016-535

100 years of piezoelectric materials in acoustics: from a sonar to active metasurfaces

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Abstract
Since the discovery of the quartz ultrasound generator by Paul Langevin in 1917, piezoelectric materials has been successfully applied to many acoustic devices, which have greatly improved our lives. Nowadays, the piezoelectric transducers can employ a vast set of piezoelectric materials such as single crystals, ceramics, polymers, biopolymers, macro fiber composites, ferroelectrets, flexoelectric materials and some others. In this Paper, a brief review of the use of piezoelectric materials in electroacoustic transducers will be given. Emphasis will be put on the modern applications of piezoelectric materials to the acoustics, especially on the method of active control of their elastic properties by means of active shunt circuits. The recent application of this method allowed the construction of so called active acoustic metamaterials (AAMM) and metasurfaces. The AAMMs based on the piezoelectric transducers offer the fabrication of efficient sound shielding structures with a low weight, a large area and a small thickness compared to the wavelength of a sound wave. It is evident that such sound-isolation structures may be applicable in devices with severely restricted weight constraints. It has been recently discovered that the great sound isolation efficiency of the AAMM is obtained in the regime of a negative acoustic impedance. Stability of the AAMM operating in the regime of a negative acoustic impedance will be analyzed and discussed.

Materials for Noise Control:
Paper ICA2016-597

Adaptive acoustic metasurfaces for the active sound field control

Kateřina Steiger[(a)]**, Pavel Mokrý**[(b)]**, Jan Václavík**[(b)]**, Pavel Psota**[(a)]**,**
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Abstract
Active acoustic metasurfaces (AAMSs) have been recently recognized as very efficient sound isolation structures, which can have large lateral dimensions perpendicular to the direction of the sound wave propagation but very short lateral dimension along the direction of the sound wavevector. The sound isolation principle of AAMSs is based on active tuning the specific acoustic impedance (SAI). This is achieved by means of active tuning of elastic properties of piezoelectric transducers, which, therefore, represent the core element of the AAMSs. Using this approach, it is possible to actively control the acoustic coefficients of transmission and reflection of AAMSs. An important point, which has been recently discovered, is the fact that the great suppression of the transmission coefficient can be achieved in the regime, when the SAI of the AAMS is negative. The function of the AAMS in varying operational conditions or in a wide frequency range, however, put delicate stability conditions on the negative values of SAI. In order to keep the AAMS in the stable operation, a concept of adaptive acoustic metasurfaces (AdAMSs) is introduced in the Paper. The methods for the real-time estimation and the active control of

the SAI values of the AdAMSs are presented. It will be shown that the accurate control of the distribution of the SAI on the surface of the AdAMS makes it possible to control the transmitted sound field not only in the magnitude but also in the direction of the transmitted sound wave.

Materials for Noise Control:

Paper ICA2016-491

Redesigning Helmholtz resonators to achieve attenuation at multiple frequencies

Nicolas Etaix[(a)], Kyle Crawford[(a)], Ruth Voisey[(a)], Hugh Hopper[(a)]
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Abstract

Helmholtz resonators can be used to reduce narrowband noise levels at low frequencies. These resonators are generally designed to target only one frequency. In order to attenuate multiple frequencies, it is possible to use several Helmholtz resonators tuned at different frequencies and located one after the other, or to connect several Helmholtz resonators in series. However, this may have implications on the package size.In this work, the exploitation of cavity modes is investigated as an alternative approach to targeting multiple frequencies from a single Helmholtz resonator. Numerical methods are used to illustrate the influence of neck location on mode selection and resonance frequency. The influence of multiple neck openings into a single cavity is also investigated. The numerical simulations are shown to be in good agreement with experiments for various designs of multi-resonant Helmholtz resonators. Finally, by way of example, a multi-resonant system is designed where the two principle resonant frequencies are tuned independently by adjusting the location of the neck and shape of the cavity. The paper highlights a design methodology to take advantage of the cavity modes within a Helmholtz resonator in order to control multiple frequencies from a single package.

Materials for Noise Control:

Paper ICA2016-490

Comparison of the acoustic behaviour of porous materials in compressed and uncompressed conditions

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Abstract

Conventional methods to evaluate the absorption coefficient of materials use either a large reverberation room or wave guides such as standing-wave tubes or impedance tubes. These last methods have recently been extended so that other material properties such as airflow resistivity can also be evaluated using the same tubes. An advantage of the impedance tubes is that they can also be used to measure the acoustical and non-acoustical properties when the materials are under compression. The current study investigates the differences between two-microphone systems and three-microphone systems, and assess both the absorption coefficient and the flow resistivity of porous materials such as rock wool and fibreglass in both compressed and uncompressed conditions. Finally, the results of the study are discussed.

Materials for Noise Control:

Paper ICA2016-589

Effect of the three-dimensional microstucture on the sound absorption of foams: a parametric study

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Abstract

Sound absorption arises mainly from visco-thermal dissipation of a pressure wave propagating through a porous material. First-principal calculations and X-ray computed tomography experiments reveal that the sound absorbing behavior of real foam samples made from dispersion of gas bubbles in liquid matrices can be directly described from a simple threedimensional regular model of hollow spheres, which opens new avenues for optimal design of acoustic materials. However, the proper choice of bubble and interconnection sizes, two critical parameters of the foam's morphology, depend on the sample thickness, the frequency range of interest (the choice of the optimization criterion), and the kind of excitation (normal or diffuse incidence). This presents a problem: a small interconnection combined with large pores is necessary for thin sample thicknesses in order to provide a resistive and tortuous layer of porous sample, but interconnections have to be larger and pores smaller for thicker samples to be able to dissipate viscous energy in all the thickness of the material by avoiding an excess of reflected waves. Moreover, the manufacturing process places constraints on the accessible range of porosities and pore radius. The selected criterion is a sound absorption average over third octave bands between 125 Hz and 4000 Hz. In this communication, massive computations are therefore used to provide guidelines for selecting the appropriate foam's morphology.

Materials for Noise Control:

Paper ICA2016-882

Characterization of mufflers

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Abstract

Mufflers are widely used to reduce the exhaust and intake noise of fluid machines for different applications. Every muffler has to be tailored carefully to the engine to which it is connected. One very important tool of muffler design is the measurement of its properties, acoustic performance and pressure drop. Introduction of flow is a key issue as it simulates a real engine situation. The two source technique has been proven to be the most stable and most efficient technique to characterize the full scattering matrix of the muffler. At several companies and research institutes, this has become a standard measurement which is repeated frequently throughout the design process. This paper describes a new platform to measure the passive and active (flow-generated noise) properties of mufflers. Stepped sine excitation is used with simultaneous excitations from both sides of the muffler. The stepped sine excitation is optimized to reduce the needed time without jeopardizing the quality of the measurement. Measurement of flow background noise, microphone coherence, and pressure drop are also performed. This platform is based on a combined JAVA/NI software and National Instruments Data Acquisition cards, to automate the measurement accounting for different theoretical and practical considerations.

Thursday afternoon, 8 September 2016 **Dr. Valsecchi Auditorium**
14:30 - 14:50
Acoustical Measurement and Instrumentation
SI1 - Sound Intensity and Inverse Methods in Acoustics

Sound Intensity and Inverse Methods in Acoustics:
Paper ICA2016-198

On the relationship between sound intensity and wave impedance

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Abstract
Following a recent paper by one of the author ["On the physical meaning of the power factor in acoustics", J. Acoust. Soc. Am. 131(1), 269–280 (2012)] where the concept of complex intensity has been fully developed from the physical point of view and its spectral properties have been highlighted, the present communication focuses on the relationship between sound intensity and wave impedance. It will be shown how the spectrum of the complex sound intensity magnitude is directly connected with the spectrum of the wave impedance in some model fields (plane quasi-stationary waves and spherical waves) so prefiguring a new methodology for measuring the sound intensity.

Thursday afternoon, 8 September 2016 **Dr. Valsecchi Auditorium**
14:50 - 16:10
Acoustical Measurement and Instrumentation
SI2 - Acoustical Measurements and Instrumentation

Acoustical Measurements and Instrumentation:
Paper ICA2016-859

Broad band air ultrasound refernce sound source

Angelo Campanella
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Abstract
New air ultrasound reference sound sources are produced to provide a steady and reliable source of air ultrasound useful for the purpose of calibrating air ultrasound microphones over the frequency range from 10 kHz through 400 kHz. The air ultrasound field is created by mechanical means. In the first source, the ultrasound emission is created by shear turbulence from the surface of a rotating cylinder. The sound field intensity at a fixed point 0.5m from the rotating cylinder oer the frequency range from 10 kHz to 100 kHz is determined with a calibrated microphone. A second source is created to operate up to 400 kHz by ultrasound emission from a small jet of releasing compressed air. The sound source intensity a a point 8 cm from the jet source is determined by the reciprocity calibration technique. Calibration data for both units will be presented.

Acoustical Measurements and Instrumentation:

Paper ICA2016-792

SAMSoft: Acoustic device automatic measurement system software

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Abstract

A common way to experimentally characterize a linear time-invariant acoustic system is by measuring its impulse response for each location of interest. Currently several methods exist for such purpose, being signal deconvolution the one which exhibits better performance. Moreover, measurement systems are usually aimed to particular applications, working with expensive platforms and proprietary software. In this paper design and development of specific software called SAMSoft, which manages an automatic measurement system for acoustic devices, are described. SAMSoft presents a modular, scalable and easy to upgrade design developed in Matlab, based on the model-view-control system pattern which enables source code reuse and simple functionality expansion. Application allows impulse response measurement with different excitation signals, time windows introduction to obtained measurements, time and frequency domain visualization, and spectral analysis in octave and one-third octave bands. It runs on a hardware based on a control unit and a mobile platform, being capable of measuring 360º in the horizontal plane with an angular resolution of up to 0.06º. Impulse response of different acoustic transducers measured showed a signal noise ratio of up to 40 dB in frequency bands under 100 Hz. In power measurements for octave and one-third octave bands a maximum error of 0.12 dB was obtained.

Acoustical Measurements and Instrumentation:

Paper ICA2016-388

Implementation and analysis of international standard for electroacoustic performance evaluation of hearing aids

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Abstract

Hearing aids are the most common device used by hearing impaired people. The hearing aids electroacoustic performance is of utmost importance for hearing care professionals to properly choose and adapt the device for a particular individual. Harmonized methods for determining the electroacoustic characteristics are fundamental to allow comparison between different hearing aids. Therefore, the International Electrotechnical Commission has published a series of standards giving recommendations for hearing aids electroacoustic measurement procedures. This article presents an analysis of the implementation of the standard IEC60118-0: Measurement of the performance characteristics of hearing aids. This standard is used to obtain results for manufacturer data sheets. First, a test setup validation was conducted by comparing the results with those measured by an accredited laboratory for a reference hearing aid. After that, investigations were made to analyze the influence of different measurement aspects, like hearing aid positioning, presence of a control microphone at the test space and differences between an ear simulator and a 2CC coupler. Comparisons with the reference showed good agreement between the results considering the respective uncertainties. The use of a 2CC coupler in IEC 60118-0 was considered the better approach from the perspective of the standard purpose. The influence of the control microphone position and presence was found to be minimum in all investigations made.

Acoustical Measurements and Instrumentation:

[Paper ICA2016-190]

Bi-lateral comparison of pistonphone calibration between INMETRO and INTI

Jorge Martín Riganti[a], Federico Ariel Serrano[b],
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Abstract

In 2001-2002, Instituto Nacional de Tecnología Industrial - Unidad Técnica Acústica - INTI (the National Metrology Institute of Argentina) and Instituto Nacional de Metrologia, Qualidade e Tecnologia - INMETRO (the National Metrology Institute of Brazil) participated in a comparison of pistonphone calibration within the Inter-American Metrology System called SIM.AUV.A-S1. However, since that time no other comparison was performed. Therefore, it was agreed between INTI and INMETRO that a new comparison of pistonphone calibration could be performed between these institutes with the aim to keep the support to the sound pressure level quantity. Each laboratory used its own procedure for the measurement of sound pressure level, frequency and total harmonic distortion. INMETRO was the pilot laboratory and the international standard IEC 60942 guidelines were followed. The institutes' results were collected throughout the project and were calculated the module of normalized error which are all less than 1, i.e. the results are satisfactory.

Thursday afternoon, 8 September 2016 **Dr. Valsecchi Auditorium**
16:30 - 18:30
Acoustical Measurements and Instrumentation
SI2 - Acoustical Measurements and Instrumentation

Acoustical Measurements and Instrumentation:

[Paper ICA2016-87]

Bi-lateral comparison of LS1P microphone calibration between INTI and INMETRO

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Abstract

In 1997-2000, the Instituto Nacional de Tecnología Industrial - INTI (the National Metrology Institute of Argentina) and the Instituto Nacional de Metrologia, Qualidade e Tecnologia - INMETRO (the National Metrology Institute of Brazil) participated in a comparison of microphone calibration within the Inter-American Metrology System - SIM called SIM.AUV.A-K1. However, since that time no other comparison was performed within SIM. Therefore, it was agreed between INTI and INMETRO that a new comparison of microphone calibration could be performed between these institutes with the aim to keep the support to the pressure-field sensitivity level. One 1-inch laboratory standard microphone designed for pressure-field (LS1P microphone) was chosen from the pilot laboratory (INTI) since its known stability and calibration history and once it was returned to the pilot laboratory it was recalibrated to certify its stability after travels. Each laboratory used its own procedure for the measurement of sensitivity level and the international standard IEC 61094-2 (1992) guidelines were followed. The institutes' results were collected throughout the project and were calculated the module of normalized error which are all less than 1, i.e. the results are satisfactory.

Acoustical Measurements and Instrumentation:
[Paper ICA2016-635]

Sound power level determinations at laboratory and field environments: An experimental comparison

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Abstract
Sound power level statement of equipment and machinery is increasingly relevant in terms of regulations and industrial competitiveness. When modelling the real world, Lw values constitute the input data for Lp predictions in diagnosis and planning of noise control strategies. Lw declaration has a major technical and economical impact, and accordingly, trying to keep the measurement uncertainties as low as possible, while balancing testing complexity, time consumption and costs is quite required. In the present work, a testing survey of real equipment emitting steady noise with different DIs and spectra is described. The applied methods had different grade of precision, ranging from ISO 3745 to ISO 3476, ISO 3741 based on Lp measurements, ISO 9614-2 based on Li , as well as a simplified proposals. The testing environments comprised an hemianechoic chamber, a reverberation chamber, a semi-reverberant normal room and an industrial hall. The calibration of a Reference Sound Source, RSS, BK 4204 [ISO 6926] used in the comparison tests is also described. A comparative analysis of results is made, considering the DI of sources, number of points, heights and environments, estimating the deviations of the different tests with respect to the reference precision method described in ISO 3745.

Acoustical Measurements and Instrumentation:
[Paper ICA2016-382]

Statistical study of the sound coverage in facade sound insulation measurement using different types of loudspeakers

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Abstract
The international standard ISO 16283-3: 2016 provides procedures to determine the airborne sound insulation of building facades, both for whole facades (global methods) and for facade elements (element methods), such as for doors, windows, etc. In both cases, the standard offers the possibility of using environmental noise as the sound source (road traffic noise, railway noise and aircraft noise) or, alternatively, a loudspeaker that emits broadband noise as an artificial sound source. When using an artificial sound source, the loudspeaker directivity requirements have been established to ensure uniform sound coverage across the facade area. This work analyzes the distribution of sound pressure level on the facades for the types of loudspeakers most commonly used in acoustic testing from a statistical point of view. This study has been carried out for both free field conditions, which is the condition specified for the qualification of loudspeaker directivity by the international standard, and actual conditions (i.e., "in situ" measurements). The purpose of the study is to determine the extent the speaker directivity requirement of the ISO 16283-3 standard guarantees correct sound coverage of facades under "in situ" measurement conditions. Finally, practical conclusions on loudspeaker choice have been derived, based on the behavior of the different types of loudspeakers applied to this type of measurement.

Acoustical Measurements and Instrumentation:
Paper ICA2016-619

A quantitative comparison of directional waveseparation techniques in tubular waveguides – An experimental evaluation

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Abstract
Acoustic pulse reflectometry (APR) has proven to be a fast and non-invasive method for inspecting ducts and pipes. A typical APR system is assembled of a loudspeaker coupled to one end of a length of tubing, termed the source-tube, with a microphone mounted on the inner wall. By injecting an acoustic pressure wave into the gas within the tube and measuring the resulting reflection sequence, it is possible to identify and characterise features within the tube. To implement a practical system, a short source-tube is preferable but results in overlap of forward and backward propagating pressure waves at the microphone location, convoluting interpretation of the reflection sequence. Acoustic wave separation can be used to overcome this issue by using multiple microphones positioned axially inside the source- tube. In this paper, two wave separation algorithms (one time domain and one frequency domain) are compared in terms of their quality of separation using a quantitative measure developed by DeScantics and Walstjin, referred to as the separation index. DeScantics and Walstjin use the separation index to assess separation quality at a given frequency. However, separation quality is frequency dependent and this dependence is addressed in the present work. The wave separation algorithms presented rely on accurate methods of determining the inter-microphone transfer functions; a number of methods are implemented in the present work. Separation quality, as a function of frequency, is presented and discussed for all of the wave separation implementations tested. Results show that optimised time domain inter- microphone transfer functions give better separation quality across the operational bandwidth than transfer functions obtained by theoretical and empirical frequency domain methods.

Acoustical Measurements and Instrumentation:
Paper ICA2016-548

Simplified two-load transmission tube measurements using an active absorbing termination

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Abstract
Current standard techniques (ASTM E2611) for measuring normal incidence sound transmission loss (nSTL) with a modified impedance tube, or transmission tube, require setting up two different absorbing termination loads at the end of the downstream tube. In this work, a modified apparatus and a new technique are proposed for non-intrusively changing the termination impedance. The standard transmission tube was modified with a downstream active termination, providing controlled variable sound absorption. The active termination allows performing standard two-load measurements, without physically modifying the passive absorbing load at the end of the tube, reducing potential measurement errors associated with the physical manipulation of the two passive terminations. Transmission loss measurements on two representative test conditions are found in good agreement with results obtained from standard passive two-load methods.

Acoustical Measurements and Instrumentation:

[Paper ICA2016-421]

Development of a measurement method for oblique-incidence sound absorption coefficient using a thin chamber

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Abstract

There have been a number of methods proposed for the measurement of oblique-incidence sound absorption coefficients. The objective of this paper is to present a novel method that utilizes the propagation mode expansion of two-dimensional acoustic field in a thin rectangular chamber. This paper is organized as follows. In the first part, measurement principle is presented. One of the greatest problem in practical is to guarantee accuracy and efficiency of multipoint measurement of the complex pressures. Thus, secondly, an elaborate prototype of the measurement system is introduced. Thirdly, numerical simulation of the measurement demonstrates the validity of the proposed procedures to extract mode amplitude from measured complex pressures. Finally, some examples of measured results are shown. In general, good agreement was observed between measured and theoretical values.

Thursday afternoon, 8 September 2016 **Cardenal Pironio Auditorium**

14:30 - 15:50

Architectural Acoustics - Room and Building Acoustics

AA6 - Concert Hall Acoustics

Concert Hall Acoustics:

[Paper ICA2016-1182]

An investigation into the Helmholtz resonators of the Queen Elizabeth Hall, London

Christina Higgins[(a)], **Raf Orlowski**[(b)], **Luis Gomez-Agustina**[(c)]

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Abstract

An investigation was conducted into the performance of the 2300 Helmholtz resonators of the Queen Elizabeth Hall, as part of acoustic design work undertaken by Ramboll for the refurbishment of the Southbank Centre’s East Wing. The reverberation in the hall at low frequencies is controlled by banks of Helmholtz resonators lining the walls. A bank of replica resonators was constructed and practical measurements were undertaken in a reverberation chamber to establish the absorption they provided, and to determine whether any repairs or adjustments are necessary. In addition, the effect of adding a layer of polyurethane foam over the neck to increase the acoustic resistance was investigated, along with the interaction between the resonators and a variable absorption system. The variable absorption system comprised two layers of acoustic curtains, which can be hung in front of the resonators. The combination of resonators and acoustic curtains appeared to provide a very effective broadband absorber suited to amplified music conditions.

Concert Hall Acoustics:
[Paper ICA2016-743]

Acoustical design of Awaza Convention Center

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Abstract
Awaza Convention Center in Turkmenbashi is designed and constructed by Polimeks Inc. to host the visiting top officials in Turkmenistan. The construction was completed in September 2015. In the form of a state pavilion, the building represents the power of Turkmenistan with its stately stature serving its valuable guests. The complex sits on a total area of 185,000 m^2 with an indoor area of 52,700 m^2, and includes 9 floors. There are two conference halls with 2,000 and 500 seats, one banquet hall with a seating capacity of 450 and another with a capacity of 250, and a press conference hall with a capacity of 130. The center also houses a 130-seat multipurpose meeting room for heads of states, a hall for signing bilateral protocols, and a meeting hall for government delegations. Additionally, there are six smaller conference halls with a seating capacity of 30 to 100 for special events, a 100-capacity reception hall and six special office rooms for the heads of states. Acoustical design of all these halls is conducted to meet acoustical design criteria limits for specific activity held in each space. Among those, the main auditorium imposed a challenge with its multi-function use. Stage pit, stage shell, stage tower, interior wall and ceiling surfaces are specifically designed to accommodate different activities including conference, concert and opera.

Concert Hall Acoustics:
[Paper ICA2016-320]

A comparison of concert halls' acoustics before and after renovation

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Abstract
The aim of the report is to introduce the results of extensive study of concert Hall's acoustics before and after renovation. The study deals with objective and subjective acoustic measurements. Objective assessment is based on the detailed analyze, measurement and comparison of acoustic characteristics before and after renovation. Subjective assessment is based on the collecting and analyzing opinions of artists, teachers, professors and audience. Reported results were obtained by a long-term thorough study of Halls' acoustics used for classic music concerts. The Halls were built more than one hundred years ago and seriously renovated in last five years. A special attention of renovators was given to the maintenance of floors, ceilings and walls constructions, decoration materials. Every renovation phase was studied in detail.

Concert Hall Acoustics:
[Paper ICA2016-761]

Acoustical analysis of Kennedy Auditorium, India

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Abstract
The study of Architectural Acoustics involves the understanding of sound build up and its propagation in a simple room. This knowledge can be further applied to larger rooms and performance areas such as concert halls, theaters and sports arenas. The basic acoustic parameters that govern the quality and intelligibility of sound for the listeners as well as the performers are needed to be evaluated. This helps the acousticians to classify a performance venue as poor or lively in terms of the spaciousness of sound. The acoustic measurements are therefore necessary to evaluate these parameters and find out the behavior of sound in certain spaces of a Concert Hall or any performance venue. This paper

presents an acoustic analysis of Kennedy Auditorium with the active sound systems in use. Further, the effect of the Building materials and their absorption properties has been explained in detail.

Thursday afternoon, 8 September 2016 **Cardenal Pironio Auditorium**
16:30 - 17:30
Soundscape
SS5 - Spatial Sound Recordings in Preserved Habitats

INVITED

Spatial Sound Recordings in Preserved Habitats:

Paper ICA2016-746

Preliminary research into the acoustic soundscape of Spitsbergen

Jerzy Wiciak[a], Dorota Czopek[b], Pawel Malecki[c], Agnieszka Ozga[d], Janusz Piechowicz[e]

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Abstract

There are some spaces in the world that are so unique and special that there is a need to investigate them deeply. Especially, because there are constant climate changes and irreversible vanishing of species or even environments. This work shows results of a sound pressure levels measurements and the analysis of ambisonic sounds recordings at several places in Spitsbergen (in the region of Svalbard archipelago). Soundscape analysis of this exceptional place was performed for summer time within a week measurements. Results for wild and partially urban spots are shown in following places: near the glacier (both mountain and sea ice), next to the birds breeding grounds, at seashore. The analysis of Longyearbyen settlement influence on surrounding soundscape is also provided.

INVITED

Spatial Sound Recordings in Preserved Habitats:

Paper ICA2016-706

Sounds of deer mating season in Bialowieza National Park

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Abstract

The deer mating season is a special sound event in the woods, associated with numerous sonorous animal sounds coming from different directions. This article presents an analysis of the sounds of rutting deer Cervus elaphus in the forests of Bialowieza National Park, a place of preserved forests of natural origin. Roars of deer during the September mating season were recorded using a SoundField microphone. An analysis of the recordings has been conducted using a custom application developed with the Matlab software. Every autumn, deer vocalizations are of particular importance for the acoustic climate in the highly protected forested areas. The lack of human communication and industrial noise make sounds of anthropogenic origin rare in the vicinity of the deer. The uniqueness of the soundscape in the Bialowieza National Park wilderness is due simply to the basic sounds of biophonic and geophonic origin.

INVITED

Spatial Sound Recordings in Preserved Habitats:

[Paper ICA2016-674]

Soundscape analysis based on ambisonic recordings executed in a primeval forest

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Abstract

The article shows the analysis of ambisonic recordings registered in the Białowieża Forest. The place is the oldest of its kind in Europe, and it is strictly protected. Within a year, hundreds of hours of ambisonic recordings have been made with respect to day and year variabilities. The Forest is included in the UNESCO World Heritage List and in the light of the new strategy adopted by the European Commission in 2011, all member states of the European Union were obliged to increase their efforts aimed at improvement of the present condition of ecosystems by 2020, with six major objectives for 2020 clearly determined. One of those objectives, which we pursue in our studies, is support at prevention of the loss of global biodiversity. Constant climate and environmental changes, as well as concern about the future generations, oblige us to preserve, or at least, to register the current state, along with the natural soundscape. The analysis of levels, spatial distribution and soundscape layers is shown.

Thursday afternoon, 8 September 2016 **Cardenal Pironio Auditorium**

17:30 - 18:10

Soundscape

SS2 - Soundscape and holistic analysis

INVITED

Soundscape and Holistic Analysis:

[Paper ICA2016-678]

A summary of the spatial construction of soundscape in Chinese gardens

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Abstract

The Chinese garden is a landscape garden style that has evolved over three thousand years. An idealized miniature landscape was created by Chinese emperors, scholars, former government officials, and merchants to express the harmony that should exist between man and nature. As an important component of the Chinese garden, the soundscape can embody the artistic conception and design skills of garden designers. This paper aims to investigate the development and strategies of soundscape design in Chinese gardens, through a systematic review of famous soundscape attractions from the perspective of spatial construction. In total, 62 soundscape attractions met the inclusion criteria, comprising specific sound sources and receiver points. The soundscape design of Chinese gardens matured in the Qing dynasty (1636 AD-1912 AD), the last dynasty in China. The majority of included soundscape attractions are located in royal and private gardens in Northern and Southern China, whereas there are few in public and temple gardens. In terms of dominant sound sources, the natural soundscape accounts for approximately 90% of the total attractions, waterscape being the main type of natural soundscape. Although the number of artificial soundscapes is smaller, they offer a wide variety of vivid sound sources, such as the temple bell, the paddle, and Chinese traditional musical instruments.

Soundscape and Holistic Analysis:

Paper ICA2016-656

A sociological analysis of soundwalk participant demographics and feedback in New Orleans

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Abstract

In an effort to address sound ordinance issues and better quantify and qualify the soundscape of New Orleans, multiple sound walks have been held in the older areas of the city over the last few years, specifically in the densest tourist areas (Vieux Carre/French Quarter and Marigny). Sound level data and participant response from several soundwalks are considered with respect to age, sex, occupation, stakeholder status, and language of descriptives used by the respondents. The results are examined and methods of analyzing soundwalk data are considered for future walks in general, and participant criteria is proposed to ensure a good sampling in densely mixed cultural environments with issues of contention.

Thursday afternoon, 8 September 2016 **Room 204**

14:30 - 15:50

Communication Acoustics

CA1 - The Technology of Binaural Listening and Understanding

INVITED

The Technology of Binaural Listening & Understanding:

Paper ICA2016-445

Exploiting envelope fluctuations to achieve robust extraction and intelligent integration of binaural cues

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Abstract

The human auditory system achieves remarkably robust communication performance, even in complex environments featuring multiple talkers, distracting noises, echoes, and reverberation. Although the neural mechanisms of this facility are not well understood, many studies point to the importance of binaural and spatial cues present at sound onset or during other fluctuations of the temporal envelope. Specifically, transient increases in the amplitude envelope appear to trigger the sampling of binaural information, independent of binaural-cue type or frequency range. This paper begins with a review of the psychophysical and neural evidence for such a triggering process, and an exploration of signal-processing algorithms that mimic and/or exploit that process. Such algorithms can be applied in two key directions of importance to communication acoustics: First, temporal envelopes are used to guide the strategic application of spatial cues in spatial sound synthesis for human listeners. Second, temporal fluctuations are used to guide the extraction of spatial cues from binaural recordings and intelligently group those cues into temporally and spatially coherent binaural proto-objects. These applications provide critical tests of the triggering hypothesis, the general role of temporal envelope fluctuations in binaural hearing, and the neural mechanisms of integrated binaural perception. Further, they provide powerful tools for the design of efficient audio communication systems and devices that interface with human participants in real or virtual spatial settings. Supported by NIH DC011548.

INVITED

The Technology of Binaural Listening & Understanding:

Paper ICA2016-253

Performance analysis of a compact spherical microphone array used for a sound-space sensing system for binaural presentation

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Abstract

Realizing communications with high presence requires the sharing of acurate sound field information among distant places. The key technologies for such systems are sensing and reproduction of the sound field information. Although various researchers have proposed accurate sound field reproduction techniques, studies related to accurate sound field sensing are few. Recently, techniques based on spherical harmonic analysis (SHA) have been focused on. To realize the techniques, several spherical microphone arrays have been developed because they arecompatible with SHA. Our proposed method, SENZI (Symmetrical object with ENchased ZIllion microphones),can acquire and reproduce accurate 3D sound-space information using a spherical microphone array without dependence on SHA, resulting in a simple single processing. The main component of the method is a rigid sphere with numerous microphones. Each input signal from the microphones on the sphere is simply weighted and summed to synthesize signals to be presented to a listener's left and right ears with their head-related transfer functions (HRTFs). Moreover, the weights can be changed according to a human's 3D head movement. Realizing this method as an actual system demands consideration of various physical factors related to the spherical microphone array, such as the microphone array size, the number of microphones, and their distribution. We thus examined how the accuracy of synthesized sound space information was affected by these physical factors. To do this, we applied real-time 252-ch SENZI system. The results of the performance evaluation indicated that acurate sound space information can be sensed and reproduced by the 252-ch SENZI system up to ca. 10 kHz.

INVITED

The Technology of Binaural Listening & Understanding:

Paper ICA2016-161

Examining auditory selective attention in realistic, natural environments with a newly designed paradigm

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Abstract

The topic of the present collaborative project (Medical Acoustics and Cognitive Psychology) is the exploration of cognitive control mechanisms underlying auditory selective attention. The aim is to examine the influence of variables that increase the complexity of the auditory scene with respect to technical aspects (dynamic binaural hearing with consideration of room acoustics and head movements) and that influence the efficiency of cognitive processing. Using a binaural-listening paradigm, the ability to intentionally shift auditory attention in various anechoic setups was tested. An anechoic reproduction fails to represent realistic listening experiences. Room acoustics and distracting sources are essential parts of a natural acoustic scene. The original paradigm is limited to present relatively short stimuli (i.e. digits). Therefore, the paradigm is extended to use longer stimuli to offer more opportunities. Spoken phrases by two speakers were presented simultaneously to subjects from two of eight azimuth positions. The new stimuli were phrases that consist of a single number word (i.e., 1 to 9) followed by either the German direction “UP” or “DOWN”. Guided by a visual cue prior to auditory stimulus onset, subjects were asked to identify whether the target number was arithmetically smaller or greater than five and to categorize the direction. Results showed generally greater reaction times and higher error rates using phrase stimuli than single word stimuli. The influence of spatial transition of the target speaker (shift or repetition of speaker’s direction in space) was similar across both paradigms. The extended paradigm is

therefore deemed suitable for studying auditory selective attention in more complex environments which can include room acoustics.

INVITED

The Technology of Binaural Listening & Understanding:

Paper ICA2016-525

Saccade-like head movements in non-human primates and implications of the binaural "acoustic flow” on spatial hearing

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Abstract

In foveate and unfoveate species, coordinated eye and head movements help improve target tracking and successful target identification. For example, auditory localization becomes more accurate in a lighted listening environment and when the head of the subject is unrestrained. However, our understanding of the neural basis of sound localization is based mainly on studies using head-restrained animals. How an animal moves its head in search of a target sound source remains largely untested. To study the head movement of marmoset monkeys, we designed a light-weight head-tracking system with an inertial sensor. RF technology enabled remote monitoring of the animal’s head movements under various listening conditions. We found that marmosets make fast head turns over large radial distances. The peak angular velocity can exceed 1000 degrees per second, faster than the speed of eye movement observed in marmosets. To understand how these fast head turns may affect spatial hearing, we simultaneously recorded the binaural signal at the ear-canal as the animal turned its head in the presence of a static sound source. The results show that the two binaural cues - interaural time and level difference – both exhibited sharp transitions during fast head movement. This suggests that, as a result of self-motion, the auditory brain may require specialized algorithms to decode the temporal patterns of binaural information. Model simulations based on spatial tuning properties of cortical neurons will be used to examine the implications of fast head movements on the neural basis of auditory space coding during fast head movement.

Thursday afternoon, 8 September 2016 **Room 204**

16:30 - 18:30

Acoustical Oceanography

AO1 - Acoustical Oceanography

INVITED

Acoustical Oceanography:

Paper ICA2016-687

Ocean acoustic tomography using a three-phased probabilistic model-based inversion scheme

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Abstract

This work presents an application of a three-phased probabilistic inversion scheme to rangedependent (R-D) environments. This model characterize and invert underwater acoustic signals in shallow water environments using Wavelet Packet Transform (WPT) for feature extraction, Hidden Markov Models (HMMs) for clustering, and a Mixture Density Network (MDN) for visualizing the inversion results via network output which describes the posterior distributions of the unknown parameters. The sequential patterns of the signals are taken into account in order to obtain a better and a more noise tolerant inversion scheme. Starting with a search space, we construct a data-set of observations, calculate the forward propagation model for each member and perform the WPT. The model initializes at random a number of HMMs with Gaussian mixtures as emission distributions and adapted to the data-set. A

hard-type assignment is performed while clustering, hence each signal belongs to only one cluster in each training epoch. A vector of log-scale posterior probabilities together with its corresponding target vector for each member of the data-set is calculated in order to adapt the MDN using the back-propagation algorithm. Here, this framework is applied in R-D environments for the estimation of the unknown parameters. For comparison reasons, we compare these inversion results to those using the Statistical Characterization Scheme (SCS) used by the authors in previous works. The major goal of our effort is to build an effective and stable model for inversions under high noise recording scenarios.

INVITED

Acoustical Oceanography:

Paper ICA2016-361

Acoustic scattering by parameterized concave surfaces using an implementation of the Kirchhoff Integral Method

Rui A. Rojo[(a)], Edmundo F. Lavia[(a)]

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Abstract

The scattering of sound waves that propagate in the ocean is mainly due to different types of inhomogeneities within seawater. This physical process is of vital importance for its wide range of applications in fisheries, oceanography, ecology and marine detection. This work addresses the calculation of the scattered field by concave objects using the Kirchhoff Integral Method. It mainly consists in solving Green's Integral Formula and applying the Kirchhoff approximation (i.e. the surface of the object is partitioned into insonified and shadow regions; the total field and its normal derivative in the insonified region are assumed to be equal to the incident field and its normal derivative, respectively, while they are assumed to be negligible in the shadow region). The well-studied case of convex scatterers has a quite straightforward solution whereas the concave case is more cumbersome. In this work an algorithm is presented to handle the interaction of plane acoustic waves with a parameterized scatterer of arbitrary concave shape. In order to verify the numerical results, comparisons with the exact high-frequency far-field solution provided by a collocation method are presented for a 2D object. Additionally, the algorithm is applied to a 3D torus insonified from different incidence directions.

Acoustical Oceanography:

Paper ICA2016-298

Acoustic scattering by prolate and oblate liquid spheroids

Juan D. Gonzalez[(a)], Edmundo F. Lavia[(a)], Silvia Blanc[(a)], Igor Prario[(a)]

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Abstract

The problem of scattering of harmonic plane acoustic waves by liquid spheroids (prolate and oblate) is addressed from an analytical approach. Numerical implementation of the exact solution of the scalar wave Helmholtz equation in an unbounded domain with Sommerfeld radiation condition at infinity and boundary conditions for the interface medium-immersed liquid spheroid is presented. A general solution which is an expansion on prolate and oblate spheroidal functions is computed with no limiting assumption regarding neither eccentricity nor sound speed and density ratios values. The implemented code is basically derived from software recently released to public use. A high resolutor level layer has been developed in Julia programming language. Predicted results have been verified for far-field and near-field results in agreement with previously reported approximate solutions. The software can be downloaded from authors' web site. This numerical implementation of the exact solution is applied in order to extend some benchmarks models of acoustic backscattering used in aquatic ecosystem research, namely, rigid-fixed, pressure-release, gas-filled, and weakly scattering.

Acoustical Oceanography:

Paper ICA2016-360

A review of theoretical models for attenuation of sound by bubble clouds

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Abstract

Attenuation of sound propagating through a bubble cloud has been primarily investigated at high frequencies, far above the bubble resonance. Near the resonance, the effective medium method has been used to study sound attenuation, by considering the bubble cloud as a single scattering object, whose internal acoustic properties are determined by a propagation wave number. This approach, however, do not consider the acoustic interaction between bubbles, which may be significant for dense clusters. Recently, the optical theory has been used to calculate the total power loss from the incident wave due to scattering and absorption by bubble clouds. This loss is directly related to the behavior of the scattered wave in the forward direction, and can be estimated utilizing a scattering model which incorporates multiple scattering effects between bubbles. However, two problems arises with this model: (1) it is unclear how to relate the sound extinction with the attenuation coefficient when the interaction between bubble is significant, and (2) the optical theory assumes that the receiver is away from the source, which may not be true for noise mitigation applications. This work aims to explore the limitations of these two models, in regard to the frequency range of applicability, density of bubbles in the cloud, and distances among source, bubbles and receiver. A review of published experimental data is included.

Acoustical Oceanography:

Paper ICA2016-338

The study of acoustic climate of the Southern Baltic

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Abstract

This paper presents statistical characteristics of seawater properties which are necessary for predicting propagation of acoustic waves in selected areas of the Baltic Sea. The statistics were elaborated based on long-term measurements of vertical distributions of sound speed, temperature and salinity, and nonlinearity parameter B/A. Nonlinear properties of the environment are considered in connection use of devices based on parametric acoustic wave generation. Special attention was paid to environmental aspects of propagation of underwater noise generated by ship operation. Statistical characteristics of the vertical distribution of sound are shown as mean values and the differences associated with statistical seasonal as well as long term changes.

Acoustical Oceanography:

Paper ICA2016-196

Advances on modelling, simulation and signal processing of ultrasonic scattering responses from phytoplanktonic cultures

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Abstract

High-frequency acoustic techniques provide remote sensing capability for plankton biomass estimation. While these techniques have been widely used to investigate zooplankton populations with frequencies below 1 MHz, few acoustical studies have been reported for phytoplankton organisms. Phytoplankton belongs to the lowest trophic level in the aquatic food chain and as a primary producer, plays an important role in the marine ecosystem. Furthermore, there are some phytoplanktonic species that can be used as biological indicators of polluted sea areas and other ones that produce harmful algae blooms affecting anthropogenic activities. Accordingly, acoustically monitoring of phytoplankton is potentially a useful technique for real time estimation of its numerical abundance. Quantitative measurements of acoustic

scattering from phytoplankton are difficult to perform due to the weak scattered intensity. Moreover, it is necessary to unmask the acoustic response of those organisms from spurious scattering generated by bubbles, suspended particles or other organisms in mixed plankton populations. In this work, advances on analysis and signal processing of acoustic scattering responses by phythoplankton are presented. At-lab measurements were performed by insonification of Skeletonema pseudocostatum cultures using a 5 MHz narrowband transducer driven by a pulser-receiver system. Additionally, backscattering cross-section of individual scatterers were computed using theoretical models for further simulation of acoustic backscattering corresponding to a large volume of randomly distributed scatterers. Good agreement was obtained when comparing simulation results with acoustic measurements.

Thursday afternoon, 8 September 2016 **Microcinema**
14:30 - 16:10
Musical Acoustics
MU1 - Music perception

Music Perception:
Paper ICA2016-243

Subjective preference of electric guitar sounds in relation to psychoacoustical and autocorrelation parameters

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Abstract
The electric guitar is a complex system composed by several elements that interact with each other, giving as a result an amplified version of the captured signal produced by an oscillating string made of nickel plated steel. Four different guitars were evaluated in a listening test by 45 people, using the pair comparison method. For the purpose of this work, a linear slide mechanism was designed to reduce the influence of the right-hand playing technique. The aim of this paper is to compare electric guitars on the basis of subjective preference in relation to psychoacoustical and autocorrelation function (ACF) parameters. According to the Pearson's correlation coefficient, it was demonstrated that the maximum value of Sharpness and the effective duration of the ACF τ_e were found to be significantly correlated with subjective preference.

Music Perception:
Paper ICA2016-227

Subjective preference of classical guitar strokes "apoyando" and "tirando" related to its harmonic components and autocorrelation function

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Abstract
Tone production of classical guitar performance is an essential part for musicians to transmit their sentimental and interpretative intentions. This work investigates subjective preferences of two common plucking techniques used by guitar players, apoyando (rest) and tirando (free) strokes. Six excerpts of classical guitar music with different tempos and range of frequency were performed using the two techniques and were recorded for the subjective tests. Two groups of subjects, guitar players and people who do not play guitar, were investigated to see if both groups evaluate the guitar timbre in different way or not. AB test was conducted with 50 persons for each group asking which technique is preferred and have more sound quality. Then the harmonic components and autocorrelation function (ACF) of each stroke were analysed to relate with the characteristics of the music program (tempo and frequency range) and subjective preferences. The effective duration of ACF is defined with the taue (τ_e) parameter. Results of the subjective test showed that harmonic content did not define preferences, but higher taue values of the ACF were correlated with a higher sound guitar quality.

INVITED

Music Perception:

Paper ICA2016-152

Effects of the mode of tempo change on perception of tempo change

Masuzo Yanagida[(a)], Ichiro Umata[(b)] and Seiichi Yamamoto[(c)]
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Abstract

Tempo is one of the basic factors in musical expression. Although there are studies on perception of tempo change, little is known about how the mode of tempo change affects sensitivity to the change. In this paper, we analyze the effects of modes of tempo change on perception of tempo change. Our analysis focuses on sensitivity to tempo change 46 participants were divided into three groups according to their musical experience and the type of playing they are used to. ((A) 15 inexperienced, (B) 21 pianists mostly playing solo, (C) 10 players of other instruments mostly playing in groups or as accompanying pianists). We compared the perception points in tempo change among three groups, with the presumption that the sensitivity to the tempo change would be higher in (B) and (C) than in (A). We used piano single tone sequences that change tempo gradually from the common initial value to the target values as stimuli. We set the initial tempo at 120 BPM, and the target tempo as either 144 BPM, 132 BPM, 108 BPM, or 96 BPM. We also manipulated the mode of tempo change ((I) linear, (II) exponential, (III) average of I and II). Participants were asked to indicate the time point of perception by pressing a key as soon as they perceived the tempo change. They were also asked to report verbally whether the perceived change was increasing or decreasing. Contrary to our presumption, the result of an ANOVA showed the time required to perceive tempo change for group (B) was longer even than that for group (A) for all the tempo progressions ($p < 0.05$).

Music Perception:

Paper ICA2016-727

Effects of lighting on impression of popular music

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Abstract

This study focused on the effects of lighting colors on impressions of popular music in an actual room, in contrast to a previous study, in which impressions of classical piano music were evaluated for colored images of a piano player generated through computer graphics. The previous study did not clarify the effect of the lighting color on the difference in impression compared with that under normal white lighting. This study investigated the impressions of six pieces of popular music in a room under LED lighting in colors of white, red, yellow, green, and blue. The results of listening experiments conducted with 10 listeners were as follows: Red lighting induces strong and dirty impressions, and blue lighting induces dark and gloomy impressions. These results are consistent with those of the previous study. In addition, the finding of the previous study that appropriate matching between the music and the lighting color enhances the impression created by the music itself was generally observed. However, the degree to which this tendency holds true was found to somewhat depend on the lighting color.

Music Perception:

[Paper ICA2016-1179]

The mechanical invariance factor in musical acoustics and perception

Akpan J. Essien

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Abstract

Acoustical and neurophysiological investigations into pitch perception repose on the Pythagorean string ratio theory of pitch interval. The validity of the theory has been denied recently on the platform of Invariance. Essien (2014) demonstrated experimentally that, contrary to established tradition in physics of sound, string *tension* is not constant but varies inversely with string length even though the oppositely-directed force exerted on the string is held constant. The finding called for complete review of theories and practices aimed to unravel the principle of the auditory mechanism. The present paper reports on the impact of a string's force of resistance to deformation on the string's vibrational frequency, spectral structure and change. Sub-lengths of a string are shown to have very little or no effect at all on a string's vibrational frequency and pitch. The data exposed to account refute the string ratio theory of pitch interval; they portray the force in a string as the mechanical parameter in control of spectral structure and pitch. Implications for future research in musical acoustics and perception are discussed.

Thursday afternoon, 8 September 2016	**Microcinema**
16:30 - 18:30	
Musical Acoustics	
MU5 - General Musical Acoustics	

General Musical Acoustics:

[Paper ICA2016-858]

Hidden influence of resonators for non-linear oscillators

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Abstract

Many self-sustained oscillators of musical interest consist of two resonators coupled by a non-linear oscillating flow control orifice. Examples include musicians playing reed woodwinds or brass instruments, for which one resonator is the player's wind-way and the other is the instrument, coupled by the reed mouthpiece or the player's lips. The voice is another example, with the sub-glottal and vocal tracts coupled by the larynx. To the extent that the coupling machinery is incompressible, the flow out of one resonator is equal to the flow entering the other. The combined effect of the two resonators is described by the sum of their input impedances. When the flow control characteristic is nearly linear, a change in the input impedance of one resonator is evident in the pressure signals in both resonators. But when the flow control characteristic is sufficiently non-linear, the effect is mostly evident on the pressure spectrum on the side where the change occurred but much less so on the spectrum on the opposite side. Several sound examples will be played and implications for musical performance will be demonstrated.

General Musical Acoustics:
Paper ICA2016-750

Antialiased soft clipping using a polynomial approximation of the integrated bandlimited ramp function

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Abstract
An efficient method for aliasing reduction under soft clipping using a piecewise polynomial is presented. Soft clipping is commonly used to model the saturating behavior of electronic musical systems such as guitar amplifiers and voltage-controlled filters used in subtractive synthesis. Saturations introduce high levels of harmonic distortion and, as such, are a major source of aliasing distortion which can lead to severe audible disturbances. The high level of aliasing distortion introduced by piecewise soft clippers can be mostly attributed to the discontinuities they introduce in the second and higher derivatives of the signal. The proposed method works by quasi-bandlimiting these discontinuities using a correction function defined as the integral of the bandlimited ramp (BLAMP) function. Due to the high computational costs of evaluating the analytic form of the integrated BLAMP function at every clipping point, a polynomial approximation is proposed instead. This approximation can be used to correct four samples, two on each side of every clipping point. Performance tests using sinusoidal signals show that the proposed method successfully attenuates aliasing components, particularly at low frequencies, by up to 30 dB with minimal computational costs.

General Musical Acoustics:
Paper ICA2016-692

Hidden melody in music playing motion: Music recording using optical motion tracking system

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Abstract
This paper shows a feasibility study of recording a sound using optical marker-based motion tracking cameras. Optical marker-based motion tracking system can record the motion of moving object using multiple high-speed infrared (IR) cameras. Recent development of the device enables capturing the detailed motions with high spatial precision of 0.01m and high sampling rate up to 10kHz. Therefore, not only the global movements of human body or handheld instruments but also the local acoustic vibrations can be recorded within the motion data, which can be transformed to actual sound radiating from the acoustic instrument. To evaluate the feasibility, several light-weight reflective markers were attached to various positions on the string instruments. Several musical excerpts were selected considering the cameras' Nyquist sampling rate. The instruments were played by professional players changing the loudness of the excerpts. The playing motions were recorded with a high-quality optical motion tracking system. Since the global motion trajectory is a relatively slow motion having the frequency component lower than 10Hz, an audible signal could be retrieved from the motion tracking data with low-pass filter. Although the current professional motion tracking system requires significantly high signal-to-noise ratio and can only retrieve the sound up to far less than 5kHz, but the result of the experiment shows that the optical marker-based motion tracking system can be useful in recording sound information from visual domain.

General Musical Acoustics:
Paper ICA2016-601
Comments on travelling wave solutions in nonlinear acoustic tubes: Application to musical acoustics
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Abstract
A common approach to modeling nonlinear behavior in acoustic tubes of variable cross-section is to use an uncoupled travelling wave solution whose profile distorts progressively - the distortion occurs due to changes in wave speed, which is a result of the nonlinearity within the system. However, these uncoupled solutions neglect a) any interaction between the waves, even in the cylindrical case, and b) any scattering effects due to changes in cross-sectional area. This paper attempts to identify what effect this separation of travelling wave solutions has compared to a coupled-wave solution. This is done with simple numerical time stepping methods for case a) to show the overall deviation of the solution when interactions are neglected. For case b) dispersion analysis is used on the linearized system to highlight the effect of scattering on the dynamics of the system. 1

General Musical Acoustics:
Paper ICA2016-207
Soprano singing, with and without resonances
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Abstract
Oscillating vocal folds are acoustically loaded by resonant acoustic ducts upstream (trachea) and downstream (vocal tract). Further, the frequency range of sopranos means that they must deal with the first resonance in each duct; indeed, they often tune the vocal tract resonance near the oscillation frequency. So, what happens when the resonances of the vocal tract (and the large impedances they produce) are removed? In this study, sopranos sang with their lips sealed around an acoustically infinite pipe, which reduces the impedance of the first tract resonance (R1) to a small, resistive load. The singers performed pitch glides over the frequency range F3-C6 (nominally 175-1049 Hz), which covers their subglottal resonance and also the range where R1 would lie in normal singing. In general, no instabilities were produced. This ability to sing into a non-resonant downstream acoustic load, and on either side of the upstream resonant load, demonstrates that a particular phase of load (inertive or compliant) is not essential for stable vocal fold oscillation.
What happens when the acoustic load suddenly changes? Later, the same singers were asked to sing a fixed pitch into the same non-resonant load while keeping their eyes closed. A plug was rapidly removed outside the lips, which introduced an acoustic reflection similar to an open mouth at the interface with the outside air, and therefore a resonant load. This abrupt change to the acoustic load on the vocal folds triggered brief instabilities in vocal fold oscillation. A return to steady oscillation after the change was possible regardless of the sign of the acoustic loading (inertive or compliant), which together with the experiment described above, suggests that stable vocal fold oscillation is possible into a wide range of acoustic loads.

General Musical Acoustics:
Paper ICA2016-482

Interval accuracy: A preliminary study of isolated choir voices singing medieval liturgical chant

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Abstract
The study and production of chorus effect has been a research and development issue during the last decades. In this paper we present a preliminary study of accuracy in choral intonation intervals. Six semi-professional male singers from a medieval liturgical chant choir are recorded individually in an anechoic chamber singing two songs belonging to the Mozarabic repertoire. Although recordings are independent, synchronization is achieved by allowing singers to listen to a rehearsal of the song and to watch the choir director on a screen while they sing. Pitch is automatically extracted from the recordings and the stable parts of the notes are marked manually. Mean pitch is calculated for each singer's stable notes and the pitch change, or interval, between consecutive notes is computed in cents. This interval is compared with the theoretical one indicated by the music sheet and the error (actual interval – theoretical interval) is obtained. Statistical analyses show that all the singers present the same median, around 0 cents, but there are significant differences in error dispersion. These results provide a new insight into the characteristics that make up the chorus effect.

Thursday afternoon, 8 September 2016 **Auditorium 2**
14:30 - 16:10
Psychological and Physiological Acoustic
PP3 - Psychological and Physiological Acoustics (others)

Psychological and Physiological Acoustics (others):
Paper ICA2016-240

Research on sound hypersensitivity: Parameters of Discomfort and Attraction

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Abstract
The purpose of this article is to collect primary data on the manifestation of sound hypersensitivity, ie., what the noises that most disturb the human ear, and which sounds are more pleasing. The idea is to analyze how the distortion of the sound perception happens, using as sampling a group of children diagnosed with Autism Spectrum Disorder (ASD), which due to medical condition, have proven sensory hypersensitivities, mainly related to sound. Not realizing the sound properly causes great harm understanding of space by humans. In addition, sensory sensitivity can be so intense that it prevents independent life in society. This research has two stages with different methodologies. In the first stage are applied questionnaires with parents or guardians of children with ASD, on their sound sensitivities and manifestation's forms of irritation or pleasure. In the second stage, the results are sorted according to Murray Schafer methodology (Schafer, 1977).The ultimate goal is to identify the main noise sources that produce the most annoying sounds and more attractive, for Children with ASD, information that has never before been researched and synthesized. And after the classification of the results, the proposal is to describe the form of manifestation of sound hypersensitivity, and what are the main factors for its occurrence. Sensitivity patterns are sought in order to subsequently propose space solutions that help a more representative group of individuals.

Psychological and Physiological Acoustics (others):

Paper ICA2016-432

Effects of background speech on reading performance in adults: Results using a new test procedure

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Abstract

Among the various factors that determine the working environment noise is often rated as stressful by many employees. This holds not only for especially noisy workplaces, with a potential of hearing damage, but also for many other settings with lower noise levels like, e.g., offices, hospitals or schools. It is known that cognitive performance can be affected by background noise, even at moderate levels. This has frequently been reported for effects of spoken language on working memory. There is less experience with the effects of moderate noise on other cognitive tasks. As reading is an essential requirement at work, a new reading task was developed and applied to investigate effects of noise. The development of the procedure is based on results from previous studies on noise effects on reading carried out by the first author. The procedure is designed as a computer-based task for adult workers with normal reading ability. The participants have to find and mark mistakes in written sentences, under a moderate time pressure. In a pilot study 12 participants worked on this task twice, once in a silence condition and once during the presentation of background speech at a moderate level. Even the data from this small group revealed that the test procedure is of high practicability. With respect to the two different experimental conditions, it was found that there were significantly more correctly finished items and less reported effort in the silence condition than under noise. The results indicate that the new procedure may have the potential to serve as an instrument for the quantification of effects of noise or specific noise characteristics on one important aspect of cognitive performance at work. Further measurements with more participants and further sound conditions will be carried out in near future.

Psychological and Physiological Acoustics (others):

Paper ICA2016-435

Auditory fMRI correlates of loudness perception for monaural and diotic stimulation

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Abstract

Loudness as the perceptual correlate of sound intensity is formed by some neural processing along the auditory pathway from the cochlea to the cortex. The loudness of a sound is largely determined by its level. Still, there are several other acoustical factors like stimulus bandwidth, duration, modulations, as well as personal factors like, e.g., the individual hearing status, that may affect perceived loudness. Binaural loudness summation refers to the finding that a binaural sound is perceived as louder than the same sound presented monaurally at the same level. Some hearing impaired listeners show an increased binaural loudness summation for broadband stimuli. The physiological background for this effect is not yet clear. We report an auditory functional MRI study comparing results from normal hearing and hearing-impaired listeners for monaural and diotic stimuli presented at different intensities. All listeners completed a categorical loudness scaling procedure, to allow for an analysis of the auditory fMRI data with respect to both, physical sound intensity as well as individual loudness perception. The results indicate systematic differences across the different stages of the auditory pathway, when comparing level and loudness-related brain activation. The brain activation is systematically increasing with sound level at all stages from brainstem to cortex. Specific effects related to individual loudness and loudness summation can be demonstrated in primary auditory cortex and in auditory association areas in Planum temporale, while the activation in more lateral regions on the first transverse temporal gyrus (Heschl) in cortex as well as in auditory brainstem structures appears to be less specific for individual loudness judgements.

Psychological and Physiological Acoustics (others):

Paper ICA2016-288

The harmonic hydro-mechanical movement of the cochlear fluid is neither a wave nor a vibration

Santos Tieso[a], Lucas Fantini[a], Francisco Messina[a], Nahuel Cacavelos[a], Gilda Farelli[a], Leonardo Zavala[a], Maria Tieso[a], Sebastian Iezzi[a], Federico Adrián Bosio[a].

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Abstract

Physical descriptions of current harmonic movements are inadequate to describe the phenomena occurring within the cochlear physiology. In this paper the hydro-mechanical harmonic movement of the cochlear fluid is described. It is possible to distinguish this type of movement as a different physic category of a vibrating solid in a mass–spring–damper system and the wave propagation in a medium. Also, it conceptually explains similarities and differences between the three types of movement mentioned. Finally, it highlights the importance of the existence of the movement in the volume perceived relating it with the different phenomena occurring in the ear.

Psychological and Physiological Acoustics (others):

Paper ICA2016-9

Perceptual consequences of auditory training in young blind persons

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Abstract

Correct interpretation of acoustically conveyed information is very important for blind and visually impaired children and teenagers in a context of safe and effective navigation in the urban environment and protection against self- and social exclusion. Characteristic features of a sound source are encoded in physical parameters of the sound it generates. The features permit distinction of different sound sources and evaluation of changing acoustic situation. The aim of acoustic training is to shorten the time necessary for execution of the auditory information processing, sensibilize the blind persons to differences in sounds, and teach them to focus auditory attention on small differences in parameters of an acoustic wave which is essential for independent and correct interpretation of environment by hearing and listening. Acoustic training had two versions: auditory training or music training. Results of both versions in blind/visually impaired children and teenagers are presented. In both trainings sounds were presented via headphones. In the auditory training the subjects had to answer questions related to presented sounds. The music training was based on passive listening to sounds. Children 7-12 year old and teenagers13-19 year old were the subjects. The auditory training was beneficial for tested teenagers. For small children the auditory training was not as effective as for adolescents. Effects of music training were ambiguous both for children and teenagers. Possible explanation of the result will be given.

Psychological and Physiological Acoustics (others):
Paper ICA2016-730

A research on understanding of piano players' perception of sonic environment and control of performance through brain function analysis and interview

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Abstract
In this paper, we are studying about the method in which we could catch the relationship between players' perception of sonic field environment and the following control of playing behavior characteristically. And, based on the above, we made some measuring experiments on some actual and performance conditions in brushing up processes, and analyzed and considered about the difference between experimental results from the view point of the perception and behavior control. Already, it has been shown that measuring the change in frontal lobe brain blood flow using Near-infrared spectroscopy (NIRS) Brain-measuring-apparatus and comparative-analyzing the measuring data was difference from a variety of psychological conditions. In this paper, adopting the above method, we measured subjects' frontal lobe brain blood flow on two actual practicing conditions of the early stage "playing at sight state" and the final stage "automated levels performance state" in proficiency-process. After some supplementary interviewing to subjects about the measuring practices, we analyzed sonic environment and control of performance behavior. The results show that we could observe each status of "the perception and behavior control" characteristically on actual performance conditions of the early stage "playing at sight state" and the final stage "automated levels performance state" in proficiency-process.

Psychological and Physiological Acoustics (others):
Paper ICA2016-612

A physically realisable lumped parameter model of the organ of Corti

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Abstract
Predictions of the active mechanical response of the human cochlea have been obtained from an elemental model, using an analytic formulation of the fluid coupling and a lumped parameter representation of the organ of Corti. The lumped parameter model is derived from the organ of Corti geometry and has three degrees of freedom, corresponding to the basilar membrane (BM), the reticular lamina (RL) and the tectorial membrane (TM). The analytic model of the fluid coupling allows the far field and near field fluid components to be accounted for separately, with the active feedback force due to the outer hair cells being proportional to the relative displacement between RL and TM. Simulate results show that the present model does reproduce in a realistic way the dynamics of the active and passive basilar membrane response in both frequency and time domains.

Psychological and Physiological Acoustics (others):

[Paper ICA2016-766]

Infant exploratory movements to localize a sound object in peri- and extrapersonal space

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Abstract

Embodied cognition approaches claim that perception is not possible without action. To perceive implies extract meaningful information from a complex environment by generating relevant sensory information contingent with current motor behavior. Since birth, the child's behavior is oriented towards an active exploration of the environment. Evaluation of infant's spontaneous responses towards sound sources, such as reaching and head turning, has an important role in the increasing interdisciplinary field of developmental psychoacoustics. Movements made by infants of 6 and 12 months old were characterized in a test of auditory distance perception. A sound object was presented to the infant in the dark in his peri (reachable) or extrapersonal (unreachable) space. The infant's behavior was recorded by two cameras mounted above and beside the participant. Micro-level analysis of video data was performed in each trial. Results showed that, even without visual cues, infants adapted his behavior to the ecological properties of the sound source: reaching was observed more frequently if the rattle was in his peripersonal space. Micro-level analysis of real-time sensorimotor organization of two infants revealed a coupling between movement (head and hand) and the sound source presentation.

Psychological and Physiological Acoustics (others):

[Paper ICA2016-150]

Dimensional approach of musical emotion recognition in relation to the running autocorrelation parameters

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Abstract

In this study, the dimensional approach of the music emotion recognition (MER) is related to running autocorrelation function (r-ACF) parameters. Therefore, a two dimensional valence-arousal plane is used to classify the emotions perceived. Ranking-based subjective tests were conducted to carry out the correlation analysis with acoustical parameters. Eight different excerpts of non-classical music were used for the test. This selection was made in order to achieve a great variety of musical genres, including Rock, Pop, Tango, Jazz, and Argentinean Folklore. The results of the analyses showed that significant correlation values were achieved between arousal dimension and the delay time of the maximum amplitude of the ACF (τ_1). A lower but still high correlation was found between the amplitude of the first peak of the ACF ($_1$) on the same dimension. Finally, valence recognition showed acceptable regression coefficients with the mean values of the width of the first peak of the ACF ($W_{\Phi(0)}$).

Psychological and Physiological Acoustics (others):

[Paper ICA2016-665]

Listen carefully: Volume adjustment behavior for portable music player usage with headphones

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Abstract

The risk of hearing damage caused by self-exposure to high sound pressure levels has increased due to the availability of portable music players (PMP) allowing users to listen to music through headphones in various everyday situations. Previous research primarily focuses on the extent and effect of high-level sound exposure, but rarely on what causes people to actively choose potentially damaging listening levels of music. The present aim is to investigate people's behavior in relation to volume control of PMP's. We take a structural qualitative approach for explorative data collection followed by an in depth discussion within the sample group. The target group for this study are young people aged 16-30, which is considered a high-risk group in the literature. Based on a clarification of the triggers crucial to the users' volume adjustments, we aim to develop a framework for designing solutions to reduce inexpedient PMP listening behavior without compromising the user experience.

Psychological and Physiological Acoustics (others):

[Paper ICA2016-279]

Pitch and rhythmic pattern discrimination of percussion instruments for cochlear implant users

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Abstract

Cochlear implants are mostly designed for speech understanding of people with hearing impairments. The appreciation of music by cochlear implant (CI) users is still under study. This work investigates the capability of CI users to recognize impulsive pitch sounds. It focuses on the discrimination of rhythmic patterns using percussion instruments samples from a kick, a snare and a ride cymbal. These signals were tuned, without changing the natural sound, into an easily recognizable pitch for CI users and were presented with different rhythmic patterns. An ABX test was carried out with two groups: 30 normal hearing (NH) subjects and 7 people with CI. The test was divided into two sessions. First, the same instrument was used for the whole pattern of each stimulus and the three instruments were compared to each other. Second, the samples of the three instruments were combined in each stimulus and the instrument of only one sample was varied and compared. Results were compared with previous studies which used the continuous tones as test signals. They showed that both groups can distinguish rhythmic differences. However NH subjects can easily recognize patterns with different sample while the CI users have much more difficulties than NH users to discriminate pitch difference in combined samples composed by different percussion instruments sounds.

Virtual Acoustics:
Paper ICA2016-519

Individual head-related impulse response measurement system with 3D scanning of pinnae

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Abstract
Head-related impulse responses (HRIRs) in the time domain, or head-related transfer functions (HRTFs) in the frequency domain, characterize the transmission between a sound source and the eardrums of a subject. They are different for each ear, angle of incidence, and also vary from person to person due to the anatomical differences. Individual measurement of HRIRs become required for applications where precise simulation of the acoustic scene is necessary, such as virtual auditory environments, and for validation of HRTF personalization methods. This presentation describes a HRIR measurement system, which uses as excitation signal logarithmic sine sweep or binary sequences known as Golay codes. The system also has a 3D scanner mounted over detachable holders that captures the digital models of the pinnae as a mesh. The results of measurements carried out in a head and torso simulator with soft pinnae are presented.

Virtual Acoustics:
Paper ICA2016-768

Assessing the contribution of binaural cues for apparent source width perception via a functional model

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Abstract
In echoic conditions, sound sources are not perceived as point sources but appear to be expanded. The expansion in the horizontal dimension is referred to as apparent source width (ASW). To elicit this perception, the auditory system has access to fluctuations of binaural cues, the interaural time differences (ITDs), interaural level differences (ILDs) and the interaural coherence (IC). To quantify their contribution to ASW, a functional model of ASW perception was exploited using the TWO!EARS auditory-front-end (AFE) toolbox. The model determines the leftand right-most boundary of a sound source using a statistical representation of ITDs and ILDs based on percentiles integrated over time and frequency. The model's performance was evaluated against psychoacoustic data obtained with noise, speech and music signals in loudspeakerbased experiments. A robust model prediction of ASW was achieved using a cross-correlation based estimation with either IC or ITDs, in contrast to a combination of ITDs and ILDs where the performance slightly decreased.

Virtual Acoustics:

[Paper ICA2016-886]

A dynamic binaural synthesis system for investigation into situational awareness for truck drivers

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Abstract

Yearly, a number of accidents happen, where cyclists are injured by right turning trucks. In Denmark, the proposed solution has been to provide a higher number of mirrors to the truck driver in order to cover visual blind spots. However, this doesn't seem to eliminate the problem. Investigations into the reason for this point to cognitive phenomena such as change blindness, where more visual information won't help. For other professional vehicle operators such as pilots, auditory solutions adding to a higher situational awareness has proven valuable.This paper describes the development of a dynamic binaural synthesis system for investigation into situational awareness for truck drivers. The system is built around several software components enabling the 3D positioning of an auditory representation of a bicycle. The sound is played back over headphones to the truck driver whose head movements are monitored and taken into account in the binaural sound synthesis. To enable experiments in real traffic, the system facilitates an operator interface where the investigator can position the auditory objects according to real bicycles appearing in the traffic.The software is organized in a number of modules communicating over a network protocol (UDP) enabling distribution on several hardware devices. The modules are: Graphical user interface, head tracking server, truck tracking, and binaural synthesis module.The function of the individual modules as well as overall topology of the system will be presented, and initial practical experience with the system used in real driving situations will be discussed.

Virtual Acoustics:

[Paper ICA2016-797]

Interactive acoustic environments for auditory rehabilitation and research

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Abstract

Hearing research studies commonly use headphones to deliver the auditory stimulus to the subject. While systems that use headphones have the advantage of being simple to construct and relatively simple to control, they typically lack cues arising from the subject's self-motion and from the user's unique head-related transfer function, both of which are important in realworld listening tasks. At the Center for Audiology, Speech, Language, and Learning at Northwestern University's Roxelyn and Richard Pepper Department of Communication Sciences and Disorders, immersive audio technologies are used to create a system that creates an interactive acoustic environment for the purpose of auditory rehabilitation and research. Because this system is physically installed in a medium-sized room, it is able to fully surround the listener in an acoustic environment with parameterized reverberation that affects all sources present in the room, including the participants' own voices. In addition, this system can also produce background noise or sound sources embedded at specific locations, or moving. This interactive environment allows natural communication, self-motion, and is compatible with all types of assistive listening devices, and is a significant improvement over headphone-based auditory environments. Use cases, performance goals and design considerations for this system will be discussed, and the range of the room's measured acoustic performance will be described.

Virtual Acoustics:
Paper ICA2016-801

Investigation into the role of the nonnegativity constraint in sound field reproduction problems

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Abstract
Given a sound field control algorithm to control the pressure and particle velocity of the field at one point located in the interior of an array of loudspeakers, it is shown that an exact solution cannot, in general, be achieved if the secondary source strength is constrained to be nonnegative. This result is put into relation with Makita's velocity vector and Gerzon's energy vector used to model human sound localisation. It is shown that, in general, Makita's vector can have the desired direction and magnitude equal or larger than one only if the non-negativity constraint is removed, and that Gerzon's vector cannot have both the desired direction and unitary magnitude.

Thursday afternoon, 8 September 2016 **Auditorium 3**
16:30 - 17:50
Virtual Acoustics
VA1 - Virtual Acoustics

Virtual Acoustics:
Paper ICA2016-681

Perception of the reverberation captured in a real room, depending on position and direction

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Abstract
Virtual auditory environments are of increasing interest, in science, but also for industrial applications. With tracking devices, the auditory scene can be explored interactively. Appropriate room simulation algorithms are essential for the plausibility and naturalness of a dynamic scene, as well as for an orientation within. It is well known, that an orientation within a purely acoustical representation of a virtual room is not easy. That especially applies to people with normal vision, who usually orientate themselves mainly by visual information. In a previous study, we investigated the ability of sighted people to distinguished four different listening positions in a shoe box-model of a seminar room. The source was kept in a constant relation to the listener to focus on reverberation. The results showed, that source directivity and training effects have a significant impact. The study presented in this paper will take a closer look at the dynamic binaural auralization of a real room with more complex acoustical properties. An assignment of listening positions within the room should be easier. Furthermore, the question for an appropriate a priori training procedure is addressed. The results will help to improve the design of virtual acoustic environments for dynamic reproduction.

Virtual Acoustics:
Paper ICA2016-170

Validating auralizations by using articulation indexes

Roberto Tenenbaum[(a)], **Filipe Taminato**[(a)], **Viviane Melo**[(a)], **Thayna Santos**[(a)]
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Abstract
In this paper, computer modeling auralizations are validated by using articulation indexes. As it is well known, one of the main challenges in validating an auralization is how to find an objective metrics to evaluate its quality. The generation of acoustical virtual reality is done by the proprietary computer code RAIOS (Room Acoustics Integrated and Optimized Software), which includes sets of artificial neural networks (ANNs) that models the head-related impulse responses (HRIRs). As output, the computer code furnishes the binaural impulse responses (BIRs) at selected positions in the room.

Convolving these BIRs with anechoic signals containing a list of monosyllables, virtual auralizations are provided. These can be applied to subjects to obtain virtual articulation indexes. On the other hand, the articulation test applied in the actual room to the same subjects provides actual articulation indexes. This is done in two university classrooms. The results show that an error lower than 2.5% is found when actual and corresponding virtual articulation indexes are compared. The conclusions are that the auralizations performed by the computer code RAIOS with the ANNs succeed and that the articulation index is a reliable metrics to validate auralizations.

INVITED

Virtual Acoustics:

Paper ICA2016-69

Parametric spatial audio recording and reproduction based on playback-setup-defined beamformers

Symeon Delikaris-Manias[a], Ville Pulkki[a]

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Abstract

Signal-independent spatial audio rendering techniques utilize recordings of spaced microphone signals, which are either linearly combined or fed directly to loudspeakers or headphones. On the other side, there are the parametric techniques, which are signal-dependent, and estimate a set of parameters that is then used for then rendering. These parameters can be for example direction of arrival or a direct/ambience decomposition of the sound field. Such systems require a low number of microphones and are very efficient in reproducing arbitrary sound scenes but they might suffer from parameter estimation errors, which leads to reproduction artifacts. A beamformer-based parametric method is shown in this work, which, instead of estimating intermediate sound-field related parameters, estimates directly the perceptually relevant parameters of the reproduction setup. The rendering setup can be either a 3-dimensional loudspeaker array or headphones. The target beamformers for the loudspeaker setup is a set of panning functions and for headphones a set of HRTFs. The system is based on two sets of beamformers for the parametric analysis and synthesis of a sound field. A set of narrow and potentially noisy beamformers is used for the analysis stage to estimate the target setup parameters such as inter-channel coherences and energies and a second set of beamformers which are noise robust is used as the source signal. The advantages of the proposed method are the reproduction of multiple instantaneous sound sources, the improvement of the single channel audio quality in the reproduction and the use of compact microphone arrays.

INVITED

Virtual Acoustics:

Paper ICA2016-826

An immersive teleconferencing system using spherical microphones and wave field synthesis

Jonas Braasch[a], Jeff Carter[b], Samuel Chabot[c] Jonathan Mathews[d]

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Abstract

Although the use of videoconferencing systems has become very common, only a few attempts have been made to transmit spatially-correct audio. One reason for this is that traditional stereophonic microphone systems cannot be used with a bi-directional transmission scheme. Because such systems are based on capturing sound sources from the far-field, their use is prone to acoustic feedback. To avoid the latter, the sound has to be captured with closely-positioned microphones or beamforming microphone system. A solution based on a spherical microphone is proposed that allows the preservation of spatial cues while avoiding acoustic feedback. The custom-built microphone consists of 16 capsules embedded in a sphere. Higher-order ambisonics is used to analyze the sound spatially and to produce beamforming patterns. The Microphoneaided Computational Auditory Scene

Analysis (MaCASA) algorithm is used to track and capture sound sources in real time. The spherical microphone can either be used as a beamformer or as a sound localization system to track participants with wearable microphones. In both cases, a wave field synthesis (WFS) system is used to reproduce the sound spatially correct. The Collaborative- Research Augmented Immersive Virtual Environment Laboratory (CRAIVE-Lab) serves as the main site for this research. The lab includes a 134-channel sound system that is used for WFS and a seamless 360-deg video projection over a floor area of 12 x 10 square meters. Two satellite labs, one containing a 64-channel WFS system and another with a 24-channel ambisonics system, exist to serve as remote sites.

Thursday afternoon, 8 September 2016 **Lounge Lateral Room**
15:10 - 16:10
POSTER SESSION - Monitor 1
Psychological and Physiological Acoustics
PP3 - Psychological and Physiological Acoustics (others)

POSTER

Psychological and Physiological Acoustics (others):
Paper ICA2016-114

Evaluation of Apple iOS-based automated audiometry

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Abstract
Audiometry has been widely used for assessing hearing situation in clinic. With the development of smart phones, several applications based on Apple iOS can be used to test audiometric thresholds automatically. However, the reliability of these tests was rarely studied in previous works. In this work, an iPhone-based automated puretone testing application was made, in which a standard procedure (STA) referring to ISO 8253-1 and a relatively quicktesting procedure (QT) were both implemented. A human behavior experiment was designed and conducted to evaluate the accuracy and efficiency of the audiometry on iPhone. Two factors were manipulated for iPhone: test environment (sound booth vs. normally quiet room), test procedure (STA vs. QT). And a conventional audiometry on audiometric equipment (Madsen, AURICAL) was also conducted as a control condition. Additionally, the test-retest reliability was also measured by repeating the 5 tests (2×2+1) on a different day. Eight young university students took part in this experiment, and they were all tested on both ears. The test orders were balanced across participants. The experimental results show 1) hearing thresholds tested in the sound booth are significantly lower than in the normally quiet room by about 3 dB HL; 2) there is no significant difference between STA and QT for hearing thresholds, but the test duration is significantly less for QT (mean 349 seconds) than SAT (mean 568 seconds); 3) there is no significant difference between the tests and retests both for hearing thresholds and test durations; 4) comparing to the results on audiometer, hearing thresholds are significantly lower and test durations are significantly less for the iPhone application. The differences observed are further analyzed and discussed.

POSTER

Psychological and Physiological Acoustics (others):
Paper ICA2016-252

Investigating the cochlear contribution to auditory pre-masking

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Abstract
Temporal masking in psychoacoustics can happen when the masker is presented before, simultaneously with, or even shortly after the probe. The last case, termed backward masking or *pre-masking*, has not been studied as extensively as forward masking and simultaneous masking. Some even suggested that pre-masking is but a matter of confusion between the masker and the probe, and a well-trained subject would not report it. Other studies showed that pre-masking can be reliably

measured within a time course of 1 to 5 ms. In this work, we present the possibility that the masker, although delivered to the ear canal later than the probe, has a chance to *catch up* with the probe once it launches a traveling wave into the cochlea. This speculation is based on the known fact that the traveling wave velocity increases against the stimulus level in the cochlea. In this work, simulation using a nonlinear model of cochlear mechanics shows that the group delay from the stapes to the 4-kHz characteristic place decreases by 0.62 ms, or longer than two cycles, when the stimulus level increases from 30 to 80 dB SPL in the ear canal. Further, due to the scaling symmetry in cochlear mechanics, the model predicts that the time course for pre-masking should lengthen if experiments are conducted at a lower frequency. The results suggest that, at least partially, the psychoacoustic phenomenon of pre-masking could be attributed to nonlinear wave propagation in the cochlea.

Psychological and Physiological Acoustics (others):
Paper ICA2016-582

Impairment of human DNA as a result of highly energetic impulse noise: Gene expression profiles as part of biological monitoring

Silvester Siegmann[a], Bernd Prisack[b], Klaus Siegmund[a], Hans Bojar[b]
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Abstract
In industry, employees are exposed to impulse noise. It is to be assumed, that cellular changes can be observed in connection to impulse noise under the influence of compression and decompression. These can either be quickly compensated by cell proteins or might require additional repair- or metabolic reactions, which should be able to be identified by altered mRNA profiles. Isolated lymphocytes, lung epithelial cells and subjects were exposed to 2 impulses within 30 minutes (Lmax~170dB) each. The samples were analysed with affymetrix chips U 95 A (12.000 genes) respectively U 133 A (22.000 genes) and the expressions were compared to the RNA-profile of a sample that was not treated with sound. No lightmicroscopic alternations can be observed. The viability does not differ als well. The comparitive analysis of gene expressions in the cell cultures shows only small varieties. The amount of altered genes is lower than it would be expected. The upregulation of individual heat shock proteins, DNA damaged induced genes, as well as the downregulation of Calmodulin associated genes and the vitamin D3 receptor is striking. In 25% of cases, 29 genes were upregulated. Out of 9.121 genes, 49 genes univariately segregate the 'temporary deaf' from the 'not temporary deaf' subjects ($p < 0,01$). However, the gene functions do not reveal direct causality to temporary deafness.

Thursday afternoon, 8 September 2016 **Lounge Lateral Room**
16:30 - 17:30
POSTER SESSION - Monitor 1
Psychological and Physiological Acoustics
PP3 - Psychological and Physiological Acoustics (others)

POSTER
Psychological and Physiological Acoustics (others):
Paper ICA2016-625

Modelling of the auditory ribbon synapse

Pablo Etchemendy[a], Ramiro Vergara[a], Manuel Eguía[a]
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Abstract
The coding of the fine temporal details of auditory stimuli by the auditory system is required for many auditory tasks. For instance, the temporal information conveys information necessary for the perception of pitch and for the angular localization of sound sources. The first stage where this kind of information is processed is the auditory periphery. Inside the periphery, the ribbon synapse (RS) of auditory inner hair cells excels for its temporal acuity, a fact that has driven many recent physiological and computational studies. In this work we present a biophysical model of the auditory Ribbon

Synapse (RS) of inner hair cells, which contains many anatomical details obtained from the electrophysiological data available in the literature, and is able to reproduce known features of the RS, namely, temporal adaptation of exocytosis due to partial vesicular depletion and gradual increment of the exocytosis rate as the membrane is depolarized. We used the model to study some aspects that are difficult to tackle experimentally, in particular, the influence of a vesicular fusion step on: (a) the formation of a "ring-like" spatial pattern of exocitosis, compatible with the spatial structure of postsynaptic receptors; and (b) the degree of synchronization of exocytosis as a function of release event size. The results described could be relevant in order to improve our knowledge of the temporal coding of auditory stimuli at the auditory periphery level.

POSTER

Psychological and Physiological Acoustics (others):

Paper ICA2016-700

Auditory environmental context affects visual distance perception

Ramiro Vergara[(a)], Ezequiel Abregú[(a)], Pablo Etchemendy[(a)], Manuel C. Eguia[(a)], Esteban R. Calcagno[(a)], Nilda Vechiatti[(b)], Federico Iasi[(b)]

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Abstract

To perceive the distance to an object through the visual modality, an observer uses a variety of cues, many of which may not be directly related to the target. This is illustrated by the fact that in a well-lit environment (with multiple visual cues) visual distance perception (VDP) is relatively accurate, whereas in a dark environment (where the observer can only see the target) VDP becomes inaccurate. Besides, a number of recent studies indicate that VDP is not only affected by the availability and reliability of depth cues, but also can be influenced by the context even in the presence of multiple visual cues. Here we provide evidence that VDP is influenced by the auditory environmental context through reverberation-related cues. We conducted VDP experiments in two dark rooms with extremely different reverberation times: an anechoic chamber and a reverberant room. We first show that the distance to a visual object located in the reverberant chamber was perceived significantly farther than the same target located at the same distance in the anechoic chamber. The results also show that the maximum distance perceived by participants correlated significantly with the perceived size of the room. In addition, participants who performed the experiment in the reverberant room reported a perceived size greater than those who performed the experiment in the anechoic chamber although both rooms are of similar sizes. Secondly we note that by separating participants between musicians and non-musicians only the former group perceived differences in the size of the room through auditory modality; moreover, only this group perceived the distance to the visual object in the reverberant chamber farther than in the anechoic chamber. On the other hand, the group of non-musicians did not perceive differences in the size of both rooms or in the perceived distance in both chambers. These results show that the auditory environment can influence the VDP, presumably by reverberation cues related to auditory perception of the size of a room.

POSTER

Psychological and Physiological Acoustics (others):

Paper ICA2016-735

Flexible Solver For 1-D Cochlear Partition Simulations

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Abstract

There is a vast literature on cochlear modelling, much of it based on theoretical and numerical analysis of the hydromechanics of the canals and the physiology and micromechanics of the organ of

Corti. During the past decades, many models have been developed from common theoretical grounds but with differences in the cochlear partition impedance, mainly because of the active mechanism adopted. Here we present a module for the Python language that allows to simulate and compare many different models in a simple manner, with the only need of writing the partition impedance expression. The outcome is a highly optimized C++ code that carries on the numerical simulation. The module can simulate models that fit in the long wave approximation of the cochlear fluid mechanics or, equivalently, a one dimensional transmission line. It allows an arbitrary number of variables for the partition impedance, including longitudinal coupling.

Thursday afternoon, 8 September 2016 **Lounge Lateral Room**
14:50 - 16:10
POSTER SESSION - Monitor 2
Signal Processing in Acoustics
SP4 - Signal Processing in Acoustics (others)

POSTER
Signal Processing in Acoustics (others):
Paper ICA2016-306

A study on near-field sound propagation based on delay-filtering for carrier and sideband waves using curved-type parametric loudspeaker

Ryosuke Uemura[a], Takahiro Fukumori[b], Masato Nakayama[c], Takanobu Nishiura[d]
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Abstract
Parametric loudspeaker is focused as a loudspeaker which has a high directivity. The parametric loudspeaker strongly emits the ultrasound modulated by the amplitude of the audible sound. The amplitude modulated (AM) wave includes the carrier and sideband waves. The audible sound is demodulated by the non-linear interaction between the carrier and sideband waves in the air. The parametric loudspeaker can propagate the audible sound in the particular area called “audible area”. However, the audible area is also formed in the rear of the target-listener. Therefore, the loudspeaker which propagates the audible sound in the near-field is required. In this paper, we therefore propose the near-field sound propagation method based on delay-filtering for the carrier and sideband waves using the curved-type parametric loudspeaker. In the proposed method, we divide the AM wave and give different delay for the carrier and sideband waves using individual-filtering. In addition, the proposed method utilizes the curved-type parametric loudspeaker to design the audible area with high power. We carried out evaluation experiments to compare the sound pressure of the demodulated audible sound with proposed and conventional methods. As results of evaluation experiments, we confirmed the effectiveness of the proposed method.

POSTER

Signal Processing in Acoustics (others):

Paper ICA2016-307

Investigation on suitable acoustic features in deep neural network for environmental sound discrimination

Sakiko Mishima[(a)], Tomoyuki Mizuno[(b)], Takahiro Fukumori[(c)], Masato Nakayama[(d)], Takanobu Nishiura[(e)]

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Abstract

Surveillance systems have been utilized for the safety of the elder people who live alone. Most of them have been utilized for detecting hazardous situations with a video camera. However, a video camera has a problem that it is difficult to monitor the dark and blind areas. In order to solve this problem, methods for hazardous sound detection have been proposed by using environmental sounds which consist of various sounds in daily life. It is important to improve the discrimination accuracy of environmental sounds in order to accurately monitor the situation. Conventional acoustic models have been realized on the basis of a hidden Markov model (HMM) with mel-frequency cepstrum coefficients (MFCCs). However, it is difficult to discriminate the environmental sound because the conventional method for constructing the acoustic model is insufficient to express acoustic features. Deep neural network (DNN) can extract complex features from input signals. High-dimensional input features are effective to input to DNN because the environmental sound has various features. However, suitable input features are required in order to reduce the computation cost. Therefore, we investigate suitable acoustic features for DNN. We employed MFCCs, mel-filter bank, and linear-filter bank as acoustic features. From evaluation experiments, we confirmed the performance of environmental sound discrimination in each acoustic feature.

POSTER

Signal Processing in Acoustics (others):

Paper ICA2016-308

An evaluation of discomfort reduction based on auditory masking for railway brake sounds

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Abstract

Railway brake sound in slowing railway vehicle causes noise problems in a station yard. Conventional noise reduction methods for railway brake sound have been proposed on basis of the improvement of brake mechanism. However, these methods have the insufficient performance of the noise reduction for railway brake sound and the railway brake sound still gives passengers discomfort feeling. In this paper, we focus on active control for reducing discomfort feeling of railway brake sound. We have previously proposed a discomfort reduction based on the auditory masking for a stationary sound. In this method, discomfort reduction is realized by emitting the noise control masker signal with the secondary loudspeaker without updating the estimated noise. However, the performance of discomfort reduction is insufficient for the railway brake sound because the railway brake sound is non-stationary noise. In this paper, we therefore propose a new discomfort reduction method based on auditory masking with the active noise estimation for the railway brake sound. We carried out subjective evaluation experiment with the actual railway brake sound. As a result, we confirmed the effectiveness of the proposed method.

POSTER
Signal Processing in Acoustics (others):
Paper ICA2016-310

Reverberation design based on acoustic parameters for reflective audio-spot system with parametric and dynamic loudspeaker

Ryosuke Uemura[a], Tomoyuki Wada[b], Takahiro Fukumori[c], Masato Nakayama[d], Takanobu Nishiura[e]
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Abstract
A three-dimensional sound field reproduction system is required in the field of the entertainment. We have previously proposed the three-dimensional sound field reproduction system using the parametric loudspeaker which represents the sound image with a high accuracy. This method can design the sound image for the target by reflecting the sound emitted from the parametric loudspeaker. This designed sound image is called “reflective audio-spot”. However, this method has a problem that the parametric loudspeaker can’t represent the room reverberation. Therefore, we have previously proposed the method using the parametric and dynamic loudspeakers to represent the room reverberation. This method controls the room reverberation on the basis of the reverberation time. However, the room reverberation has not only the reverberation but also various acoustic parameters. These acoustic parameters are important for the human perception at room positions. In this paper, we therefore propose the reverberation design based on acoustic parameters for the reflective audio-spot system using parametric and dynamic loudspeakers. In the proposed method, we utilize the reverberation time and the direct-to-reverberant ratio to perceive the sensation of the listener position in the room. We confirmed the effectiveness of the proposed method through the evaluation experiment.

Thursday afternoon, 8 September 2016 **Lounge Lateral Room**
16:30 - 17:30
POSTER SESSION - Monitor 2
Signal Processing in Acoustics
SP4 - Signal Processing in Acoustics (others)

POSTER
Signal Processing in Acoustics (others):
Paper ICA2016-314

Control signal design for reducing discomfort of infant cry based on mitigation of time fluctuation

Aomi Kobayashi[a], Takahiro Fukumori[b], Masato Nakayama[c], Takanobu Nishiura[d]
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Abstract
Infant cry may become recognizing as a noise and also cause of noise trouble between neighbors and passengers. It is also known infant cry could be as stress components, particularly mothers. Therefore it is necessary to reduce discomfort of infant cry. We have previously proposed the method which reduces discomfort of fan noise and infant cry on basis of auditory masking which is caused by auditory characteristic of human. In the previous method, discomfort reduction is realized to mask detected spectral peaks which are discomfort components of infant cry. We, however, confirmed that infant cry is a noise which has not only spectral peaks but also large and sharp time fluctuation from the feature analysis. That is to say there is a possibility that we can reduce further discomfort of infant cry if time fluctuation of infant cry is mitigated. We therefore proposed the method which reduces discomfort of infant cry on basis of mitigation of time fluctuation. Control signals which can mitigate the large and sharp time fluctuation are designed by weighting original sound sources in time domain.

There are three kinds of weighting factor based on exponential, linear and quadratic functions. Reducing discomfort of infant cry is realized by hearing infant cry and the control signal at the same time. From the subjective evaluation experiment, we confirmed the effectiveness of the proposed method.

POSTER

Signal Processing in Acoustics (others):

Paper ICA2016-614

Cross-frequency coupling and phase synchronization in nonlinear acoustics

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Abstract

The Cross-Frequency Coupling (CFC) characterizes the correlation between the phase/amplitude of a low-frequency band with those of a high-frequency band within or across time series. The nonlinear interaction between different frequencies in the form of CFC has been observed in a variety of systems: earth seismic waves, stock market fluctuations, pulsatile hormone secretions and in the scale-free activity of the human brain. In this work we experimentally explore the CFC phenomenon in the context of nonlinear acoustics. Specifically, experiments were conducted using a liquid-filled spherical chamber arranged in an experimental setup tailored for characterizing the acoustic field developed in typical nonlinear dynamics found in bubbly liquid media, like single/multi-bubble sonoluminescence (SL). The measured time series data were analyzed using signal processing algorithms specialized for quantifying CFC and phase synchronization phenomena: Comodulogram, Time-Locked Plot and Phase Locking Value. Our results suggest that in absence of bubbles, the excitatory-inhibitory interaction among the acoustic modes of the spherical resonator, coupled via 1 : n internal resonances, produces the nonlinear harmonics (N f0) observed in the system response. Besides, we interpret this experimental observation in terms of a previously reported canonical model for generating CFC in neural systems. On the other hand, it was found that the interaction between the acoustic modes of the spherical resonator with the SL bubble(s) acoustic emission, generates nonlinear harmonics (N f0) amplitudemodulated by the phase of the fundamental driving frequency (f0) in which the overall modulation strength and its variance across multiple time series data depend on the spatial stability of the SL-bubble(s).

POSTER

Signal Processing in Acoustics (others):

Paper ICA2016-881

Three-dimensional spatial sound-image design based on separating emission with curved-type parametric loudspeaker

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Abstract

A parametric loudspeaker has a sharper directivity by utilizing an amplitude modulated wave which integrates the carrier and sideband waves (ultrasound), and can form a spatial narrow audible area. Moreover, it can design the three-dimensional (3-D) sound-image on a reflective object with the reflected sound. In this paper, we propose the design method of the 3-D spatial sound-image without the reflective object. We focus on the radiation characteristics of acoustic waves which are radially transmitted from a sound source. In the proposed method, we design the 3-D spatial sound-image by reproducing the radiation characteristics of the virtual sound source in the air. Specifically, we form a focal point of emitted sounds by utilizing multiple parametric loudspeakers arranged on arc, and

design the virtual sound source on it. In this paper, we utilize the separating emission of the carrier and sideband waves to achieve the silent area from the parametric loudspeakers to the focal point. In the separating emission, the audible area is formed in the particular area where the carrier and sideband waves overlap. Therefore, the proposed method can design the spatial sound-image with the silent area from the parametric loudspeakers to the focal point. In addition, the formed audible area is too small because of too sharper directivity in the separating emission using the conventional parametric loudspeaker. Therefore, we utilize the curved-type parametric loudspeaker which has the curved surface arrangement of ultrasonic transducers and has a wider directivity. Finally, we evaluated the effectiveness of the proposed method. As a result of the evaluation experiment, we confirmed the proposed method is effective for the 3-D spatial sound-image design.

Friday, 9 September 2016

Friday morning, 9 September 2016 **Juan Pablo II Auditorium**
09:00 - 10:40
Architectural Acoustics - Room and Building Acoustics
AA7 - Isotropy and Diffuseness in Room Acoustics

INVITED
Isotropy and Diffuseness in Room Acoustics:
Paper ICA2016-863

Evaluation of isotropy of sound field in a room based on the decay-canceled sound intensity

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Abstract
Isotropy is one important element for evaluating diffuseness of a sound field. Later reflected sounds refer to reflected sounds with a longer delay of more than 80 ms. In particular, reflected sounds with a very long delay are called “reverberation tails.” Thus far, it is not clear whether a sound field has characteristic arrival directions for the reverberation tail as the reverberation tail shows not significant characteristics owing to diffusion. The purpose of this study is to develop a technique for analyzing the spatio-temporal structure of reflected sounds in the reverberation tail in order to clarify whether there are significant characteristics on arrival directions of reflected sounds in the reverberation tail. In this study, the decay-canceled instantaneous sound intensity (DC-II) is proposed for evaluating the isotropy of a sound field, especially in the reverberation tail. The C-C method, which has been developed by the authors [Proc. INTER-NOISE 2008, T. Hanyu], was employed to measure the instantaneous sound intensity. We investigated the DC-II in several sound fields. As results, the DC-II showed that sound fields have significant characteristics on the arrival directions of late arriving reflected energy, namely the reverberation tail.

INVITED
Isotropy and Diffuseness in Room Acoustics:
Paper ICA2016-578

A wavenumber approach to characterizing the diffuse field conditions in reverberation rooms

Mélanie Nolan[(a)], Efren Fernandez-Grande[(b)], Jonas Brunskog[(c)], Antoine Richard[(d)], Cheol-Ho Jeong[(e)]
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Abstract
This study proposes a wavenumber analysis method for evaluating the diffuse field conditions in a reverberant space. The wavenumber (or angular) spectrum, which results from expanding an arbitrary sound field into a plane-wave basis, is used to characterize the spatial properties of the observed sound field. Subsequently, the obtained angular spectrum is expanded into a series of spherical harmonics, and the multipole moments from this spherical expansion are used to characterize the wave field, in terms of both isotropic conditions and phase distribution. The paper examines the validity of the method and investigates how the results relate to the existing theory for the diffuse sound field in a reverberation room, based on Waterhouse’s random wave model. The analytical

framework is presented, and the proposed methodology evaluated numerically based on simulated measurements using a spherical microphone array.

INVITED

Isotropy and Diffuseness in Room Acoustics:

Paper ICA2016-527

Kurtosis as a diffuseness measure

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Abstract

This study presents a kurtosis analysis of room impulse responses as a potential room diffuseness measure. In the early part of an impulse response, sound pressure samples do not constitute a Gaussian distribution due to the direct sound and strong reflections. Such deterministic reflections are extreme events, which prevent the pressure samples from being normally distributed, leading to a high kurtosis. As the reflections are sparser and stronger, the sound field becomes less diffuse and the kurtosis systematically increases, indicating that it can be used as a diffuseness measure. The kurtosis converges to zero, as the reflection overlap becomes heavier, which is an important condition for a perfect diffuse field. Two rooms are analyzed. A small rectangular room shows that a non-uniform surface absorption distribution tends to increase the kurtosis significantly. A full scale reverberation chamber is also tested with many different diffuser settings. Results show that the kurtosis from a broad band impulse response has a good correlation with the equivalent absorption coefficient according to ISO 354.

INVITED

Isotropy and Diffuseness in Room Acoustics:

Paper ICA2016-624

Investigation across different conditions of room diffusivity measured from a variable acoustics facility

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Abstract

While a variety of objective metrics have been proposed to quantify diffusivity within rooms, the link between perception of diffusion and these metrics has yet to be fully established. In this project, impulse response measurements from numerous diffusive room conditions were recorded in the Mocap variable acoustics facility at Columbia College Chicago, which contains 1200 square feet of reversible absorptive/diffusive panels covering the walls. By configuring these panels in various orientations, two independent tests were conducted. In the first test, a direct reflection was recorded off of a 16 by 8 foot test section, which was progressively changed from fully absorptive to fully diffusive. In the second test, diffusers throughout the entire facility were arranged in multiple coverage percentages and in three unique arrangements. A number of acoustic metrics were calculated in each of these room states for comparison, including several proposed to quantify room diffusivity. In addition, binaural impulse responses using a KEMAR head and torso simulator were taken and used to generate auralizations for subjective testing to discern how perceptible the changes in diffusive conditions are for both a direct reflection and a full room response. Results from both the objective metric comparison and subjective testing are presented.

INVITED

Isotropy and Diffuseness in Room Acoustics:

[Paper ICA2016-556]

A rigorous definition of the term "diffuse sound field" and a discussion of different reverberation formulae

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Abstract
Often, Sabine's and other reverberation formulae are applied without really knowing whether the sound field is sufficiently diffuse. In this rather didactical paper a rigorous definition of the crucial term 'diffuse sound field' is proposed and the relationships to the necessary surface conditions, especially scattering, are analyzed. Also the reasons for the differences between Sabine's and Eyring's reverberation formulae are analyzed. Some other approaches for partially diffuse sound fields, e.g. Kuttruff's formula, are discussed. Some numerical investigations are added The aim is to find reliable definitions of conditions for at least an approximately diffuse sound field.

Friday morning, 9 September 2016 **Juan Pablo II Auditorium**
11:00 - 13:00
Architectural Acoustics - Room and Building Acoustics
AA7 - Isotropy and Diffuseness in Room Acoustics

INVITED

Isotropy and Diffuseness in Room Acoustics:

[Paper ICA2016-703]

Relation between acoustic measurements and the perceived diffuseness of a synthesised sound field

Michael P. Cousins[a], Stefan Bleeck[a], Frank Melchior[b], Filippo Maria Fazi[a]
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Abstract
This paper describes an investigation of different objective metrics for predicting the perceived diffuseness of reproduced sound and why common metrics, such as Interaural Cross-Correlation Coefficient (IACC), sound pressure level uniformity, and the diffuseness calculation used in Directional Audio Coding (DirAC), may be less appropriate for analysing the perceived diffuseness of a reproduced field than they are for architectural acoustics applications. A listening test was conducted to elicit the perceived diffuseness of sound fields of uncorrelated pink noise signals replayed over 19 different loudspeaker arrangements. Listeners rated how diffuse they perceived each stimulus. A range of different measurements of the sound field were then compared to the subjective test results. The data show that objective metrics do not always correlate well with the perceived diffuseness, especially for specific loudspeaker arrangements. Possible explanations of these results are discussed.

INVITED

Isotropy and Diffuseness in Room Acoustics:

[Paper ICA2016-777]

Non diffuse sound field in the reverberation room

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Abstract
The measurement method for diffuse field sound absorption (ISO 354) is troubled with low reproducibility, far worse than can be accepted in respect to design of spaces, control of quality and legal security. These differences are expected to be caused by insufficient diffusion in the reverberation room. Already geometrical modelling of sound in a reverberation chamber shows how the result of the absorption measurement depends on the scattering of reflections on walls and ceiling.

Measurements in the chamber show the increase in absorption by adding diffusing materials on the walls. Attemps have been made to describe the diffusitivity of the sound field based on the variation of the recorded sound decays. Suggestions will be given to increase the measured sound absorption, but also a method will be presented using a reference absorber to overcome the issue of different diffuse field properties in different reverberation rooms.

INVITED

Isotropy and Diffuseness in Room Acoustics:

Paper ICA2016-717

Decay curves in coupled, reverberant spaces

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Abstract

This study investigates the effect of panel and boundary diffusers in a reverberant space. Diffusers are usually mounted in a reverberation chamber to increase the diffuse sound field as recommended in Annex A of ISO 354. The ISO is not specific about the location or the material of the panels; the standard only states that the absorption coefficient of a highly absorbing material will increase and approach a maximum value. This value is usually much higher than 1 when diffusers are added. It is also known that the reproducibility of absorption coefficient measurements in reverberation chambers is unsatisfying. This study investigates the effect of panel diffusers, in particular considering that their dispositioning in a room can create coupled spaces, decreasing the effective volume of the chamber, and leading to an overestimation of the absorption coefficient. The decay curves are measured in a small chamber with panels placed in a corner creating a coupled space. Both in the empty room as well as with A = 3 m2 absorbing porous sample on the floor, the decay curves are evaluated. Additionally, the effect of boundary diffusers is considered. The decay curves for different room configurations in the occupied (with high absorption on the floor) and unoccupied state (without any absorption) are compared. The decay curve in the occupied state without any panels or boundary diffusers has a breakpoint where the slope changes its value. This can also be observed in the unoccupied state with panels placed in the corner of the room.

Isotropy and Diffuseness in Room Acoustics:

Paper ICA2016-919

Effectiveness of reverberation room design: Room size and shape and effect on measurement accuracy

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Abstract

The reverberation-room method, which assumes a diffuse sound field, has long been used for various standardized room-acoustical measurements – i.e. absorption coefficient, source power level, transmission loss. However, unsatisfactory opinions regarding the accuracy of the method, especially at low frequencies, have been reported over the years. This might be due to a deviation from the assumed diffuse-field concept, which is very challenging to implement from an application point of view. To investigate the problem and find a solution, a number of reverberation rooms of different sizes and shapes have been studied; their capacity to approximate a diffuse sound field is analyzed by means of descriptors like cut-off frequency, spatial uniformity of sound pressure and reverberation time, degree of linearity of temporal decay curves. Results obtained with the help of a numerical finite-element-based modal approach are discussed; in particular, the effect of different room sizes and shapes on the measurement accuracy is explained. Based on these findings, recommendations are proposed regarding the sizes and shapes of reverberation rooms that will give better field diffuseness and, hence, better measurement accuracy.

INVITED

Isotropy and Diffuseness in Room Acoustics:

[Paper ICA2016-277]

Correction algorithm for sound scattering coefficient measurements

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Abstract

Scattering coefficient measurements according to ISO 17497-1 are very sensitive to temperature and humidity changes in the reverberant room. Even variations as small as 0.1 K can significantly influence effective reverberation times and lead to wrong results. This phenomenon puts quite stringent limitations on how scattering measurements can be performed. This article elaborates on these limitations and discusses precautions that need to be taken in practical situations. Furthermore, we verify to what extent a stretching algorithm can help to recalibrate impulse responses and improve the quality of measured data in case measurement sequences have been subject to moderate temperature variations.

Isotropy and Diffuseness in Room Acoustics:

[Paper ICA2016-225]

Spatial variations of the mean free path in long rooms: Integration within the room-acoustic diffusion model

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Abstract

Over the last years, some extensive works have been dedicated to the modelling of the reverberant field in rooms as the solution of a diffusion process. In this model, the main parameter is the so-called diffusion coefficient of the room. It is theoretically proportional to the room mean free path, but some recent publications showed that this coefficient is not constant along an elongated room. The present work proposes a local definition of the mean free path of the room. This definition is applied to particle-tracing simulations of the reverberant field in long rooms, by using the statistics of the collisions of the particles on the walls. The results indicate a variation of the mean free path along the room, leading to spatial variations of the diffusion coefficient that are in agreement with some direct measurements of this coefficient. The proposed approach is validated for different absorption coefficient and scattering coefficient values.

INVITED

Soundscape, Psychoacoustics and Urban Environment:

Paper ICA2016-550

Soundscape standard under development

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Abstract

For more than 15 years the soundscape approach in understanding sound as a resource and not a waste is documented in research but also in COST projects. Moreover, in 2014 the first ISO Standard in Soundscape IS0 12913-1 , 2014 Acoustics –Soundscape- Part 1: Definition and conceptual framework was published. Since then discussions on further development started, and the current ISO WD 12913-2 shall bring more information and confirmation about “Data Collection” from which it is expected to provide more about the character of the holistic of Soundscape and consequent judgments. In addition, there is much new work on ANSI standards that consider life in park and wilderness areas. All of these engagements are directed to enhance the quality of life not only for humans but also for non-human beings. Such procedures can guide the process of designing our acoustic environment based on the participation of people involved. But it also forces to discuss on a level that takes into account the different perspectives of a satisfying procedure that will overcome restricted views on noise perception.

INVITED

Soundscape, Psychoacoustics and Urban Environment:

Paper ICA2016-528

Do we need psychoacoustics within soundscape?

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Abstract

For the evaluation of soundscape the perception by the human being with respect to the auditory sensation is very important, which is also influenced by expectation, experience and context. The well-known A-weighted sound pressure level is a suitable predictor at higher levels to estimate a possible damage of the human hearing by sound. But at lower sound pressure levels annoyance cannot be sufficiently predicted by the A-weighted sound pressure level. Especially in complex sound situations with different spatially distributed sound sources the human hearing is able to focus on single sound sources and to judge the sound quality depending on loudness, sharpness, roughness, fluctuation strength separately. This means that for the acoustical investigation of soundscape the use of binaural recording and psychoacoustic analysis is strongly recommended, which is currently described as informative in the new working item proposal for ISO 12913-2.

INVITED

Soundscape, Psychoacoustics and Urban Environment:

[Paper ICA2016-856]

A comparison of soundscape evaluation methods in a large urban park in Montreal

Daniel Steele[a], Edda Bild[b], Cynthia Tarlao[c], Irene Luque Martín[d], Jorge Izquierdo Cubero[e], Catherine Guastavino[f]

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Abstract

Combining surveys with other methods like observations can offer a more holistic understanding of participants' experience, with respect to activity and the evaluation of acoustic environments. Reconciling data from multiple methods remains a challenge for soundscape research, even in well-studied park settings. We compare 3 methods (behavioral mapping (n=84), questionnaires (n=41), sound recordings) to research the interaction between park users and their soundscapes over 4 sessions in the summertime. We collected soundscape ratings (SSQP, restorativeness, appropriateness) and free-format verbal descriptions, together with demographics, activity data, and personality measures. Annotated sound recordings for each observation session were compared against source and activity descriptions and free format verbal descriptors were classified into emerging themes. Within categories of sound sources, we observed different valences (e.g. within bird sounds, ducks were positive, seagulls negative.) Soundscape chaoticness was observed to vary over a small location. Wide variations in sound source identification across activity zones and across participants and researchers reveal an influence of the data collection method. Importantly, this project serves as a baseline against which we can compare soundscape studies taking place in other contexts and will inform future methodological efforts.

INVITED

Soundscape, Psychoacoustics and Urban Environment:

[Paper ICA2016-532]

Identification and separation of noises with spectro-temporal patterns

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Abstract

Acoustic signals often contain perceptually detectable noise patterns with spectro-temporal structures, causing sensations like roughness (due to modulated signal components) and tonality. Technical sounds or environmental noises are often composed of several such components. It is assumed that perceptual evaluations of such complex scenarios show larger deviations because test participants concentrate on different components depending on their preference. Therefore, it is desirable to identify and possibly separate these components allowing for an investigation of each individual noise pattern. The goal is to recognize the composition of all components corresponding to their pitch and modulation rate. Such information could be used for further development and improvement of calculation methods for psychoacoustic parameters. This paper presents different approaches based on time-frequency analyses as well as on the hearing model of Sottek evaluating a three-dimensional autocorrelation analysis as a function of time, lag, and frequency band. The extension to the third dimension allows for a better consideration of modulated signals.

INVITED

Soundscape, Psychoacoustics and Urban Environment:

Paper ICA2016-791

A novel auditory saliency prediction model based on spectrotemporal modulations

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Abstract

Previous studies indicate that soundscape perception and appraisal are influenced by the sounds that people hear and pay attention to. Hence, a model that evaluates instantaneous human attention to environmental sounds would be very useful in soundscape research. Attention is triggered by the saliency of a sound within its context. Therefore, we propose a model for predicting saliency of sounds based on dynamic modulation ripples – simultaneous modulations in the frequency and time domain. These ripples exhibit direct response in the auditory cortex of the human brain. Our model contains three stages. In the first stage, the incoming sound signal is demodulated similarly to the early stages of auditory processing, and afterwards it is correlated with each of the modulation functions of the ripples. The obtained ripple features enable the model to detect salient changes that are not accompanied by changes in more commonly used spectrogram features. We demonstrate this by comparing the model output for sound signals with the same amplitude but randomized phase spectrum. The second stage of the model integrates ripple features over time to simulate excitation and inhibition processes happening along neural pathways. In the final stage, spectral saliency is aggregated to an overall saliency using supervised training on sound environments with embedded salient sounds. We evaluate the model with a collection of natural sound fragments previously used in an EEG experiment on attention and illustrate its application in complex environmental sound scenes.

Friday morning, 9 September 2016 | **Dr. Valsecchi Auditorium**

11:00 - 13:00

Soundscape

SS3 - Soundscape, Psychoacoustics and Urban Environment

Soundscape, Psychoacoustics and Urban Environment:

Paper ICA2016-609

Accuracy of computer simulation software using hybrid models for microscale urban environments

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Abstract

To investigate environmental acoustics in large urban areas, macroscale simulation software uses calculation methods based on simplified algorithm models. However, in order to examine microscale acoustics environments such as a street or a square, those simplified algorithms may not be enough. The technological advances of the last decades in acoustic simulation software based on hybrid calculation methods, such as raytracing and image source, now allows new experimentations in urban environments. Several researches have shown the reliability of results of hybrid models when compared to in-situ measurements in closed spaces. Hybrid calculation models may also be used to simulate small open urban environments, however there are few studies showing the reliability of the results. This research aimed to investigate the accuracy of hybrid computer calculation models in microscale urban spaces. An open space with an "L" shaped edification was selected in order to provide proper reflections for the study. Acoustical measurements *in situ* were done using the method of impulse response. Computer models were also created using software Odeon v.13. Accuracy was evaluated comparatively using JND values of acoustics parameters as reference. Analyzed parameters were T30, EDT and SPL. Energy-Time curves and Impulse Responses were also compared. It was found that parameters have a good agreement between simulated and measured results, especially in mid-high frequencies. There is also a position dependent variation in T30 due to the detachment of the building and approximation to the free field. Results showed that hybrid models based software can be successfully used in the acoustic characterization of microscale urban environments.

INVITED

Soundscape, Psychoacoustics and Urban Environment:
Paper ICA2016-222

Assessment of combined environmental sounds

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Abstract
Predicting human evaluation of complex acoustical environments with different noise sources still poses a challenge. In order to explain annoyance reactions to combined sounds, various models have been developed, often based on a weighted combination of loudness or sound pressure levels. However, these models are limited regarding the evaluation of sound quality and pleasantness. Natural sounds, for example, can also have a beneficial effect in noisy environments, although they represent additional sound sources, which increase the overall sound pressure level. Behind this background, it was investigated how pleasantness ratings of singular sounds affect the overall evaluation of their respective combinations, to gain deeper insight into fundamental evaluation mechanisms. Based on the results of two listening studies, a linear regression model was proposed, which explains well the overall pleasantness evaluation of two and three combined environmental sounds, using the weighted sum of the singular pleasantness ratings and their interaction. In this model, unpleasant sounds receive a greater weight compared to pleasant ones, which presumably is due to negativity dominance and partial masking effects. Since pleasantness judgements are often confounded with loudness, another listening experiment under laboratory conditions was conducted, to separate these two variables. A positively and a negatively perceived sound, each one with three different loudness levels, were combined in pairs in all possible configurations and evaluated by test participants. The results of this experiment were used to validate the existing regression model, and they explain the interaction of pleasant and unpleasant sounds independent from loudness.

Soundscape, Psychoacoustics and Urban Environment:
Paper ICA2016-720

Soundscape identification in terms of urban morphology

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Abstract
As urban morphology decides all components of a sonic environment either in a view of physical or social aspect, it is implied to have a strong relation with a soundscape. Based on such acknowledge, this study is then going to explore different urban morphologies and their relation with soundscapes. The study chose several places with various urban morphologies in the Shenzhen China as study cases. Through field studies in these places, a series of sound recordings have been obtained, and a relationship of the place’s soundscape with its urban morphology has been systematically studied. Eventually, a soundscape is supposed to be identified by understanding its relations with urban morphology elements.

INVITED

Soundscape, Psychoacoustics and Urban Environment:

Paper ICA2016-263

Noise in a German operation theatre: Strain for the medical staff

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Abstract

Noise research in hospitals focuses mainly on the harmful effects on patients. But at least in intensive care units and operation theatres, also the medical staff is exposed to high levels of noise during considerable portions of working time. During operation sessions lasting from 30 min. to several hours, in the literature reported average Leq values range from 58 dBA to 72 dBA with maximum levels above 105 dBA. A first goal of our study was the careful measurement of the acoustic situation at typical points of exposition of the operation theatres concerned. These measurements were accomplished during one period by at least in each case 72 hours on three following weeks. For collection that subjectively noticed demand by noise during the tour was sketched its own questionnaire for the evaluation of the current work shift, in which the medical staff should judge possible acoustic factors and of the subjectively felt effects. Because of all measuring points the average values relatively close together in a range from 60 dBA to 65 dBA, and also the mediane are in a narrow range from about 60 dBA to 63.5 dBA. The two similar operation theatres 1 and 2 hardly show differences in their values; the average values of scarcely 65 dBA lie in normal ranges. The measured reverberation times were very bad for the communication. The noticed „noise at work" correlates highly with „the annoyance by noise" and the factors „communication" and/or „concentration by noise disturbed". More than one third of the asked medical staff (35 of 99) cannot exclude at least noise-conditioned errors.

INVITED

Soundscape, Psychoacoustics and Urban Environment:

Paper ICA2016-119

A comparative study on the soundscape of public spaces in coastal and inland cities in northern China

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Abstract

In order to determine the differences of the acoustic environment awareness of the general public in the coastal public space and the inland public space of China, the coastal city Huludao and inland city Harbin were selected as the research locations. In each city, objective measurement and subjective questionnaire survey were conducted in three typical public spaces. The results show that the general public of the two cities have similar acoustic sensitivity and demand for acoustic environment; the primary difference in acoustic sources is sound of water, wind and traffic; and the regression analysis indicates that there is no significant correlation between the overall satisfaction on acoustic environment and A-weighted sound level, and the overall satisfaction of the coastal public space is generally higher than that of the inland space.

Soundscape, Psychoacoustics and Urban Environment:
Paper ICA2016-75

Fountains as sound elements in the design of urban public walks soundscapes

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Abstract
One of the most common types of soundscape design are urban public walks, these are normally surrounded by buildings and avenues with common city sound and noise sources which must be minimized in order to reduce their negative effects, so that people can achieve an isolated space and time that permit relaxation and recreation during the walk. This work analyses the case of the Santa Lucia Riverwalk (Paseo Santa Lucía), in Monterrey, Mexico, an artificial riverwalk of approximately 2.5 Km, opened in 2007 in the context of the "Universal Forum of Cultures Monterrey 2007", which connects down town of Monterrey city with Fundidora Park. This walk is one of the major tourist attractions of the city and through various strategies a proper and pleasant soundscape is obtained. The strategies used in the soundscape design of the riverwalk are described, the acoustics of fountains are briefly discussed, and the uses of fountains to help masking of urban noise are analyzed. Moreover, the results of sound levels measurements produced by the fountains along the riverwalk are presented. Lastly, the aspects that must be considered in the use of fountains to mask noise in urban walks are discussed.

Friday morning, 9 September 2016 **Cardenal Pironio Auditorium**
09:00 - 09:40
Noise: Sources and Control
NS4 - Materials for Noise Control

Materials for Noise Control:
Paper ICA2016-911

Evaluation of the acoustic performances of metal powder based materials

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Abstract
Electrodes in Molten Carbonate Fuel Cells (MCFC) are constituted by porous metal mixtures. Metals employed are nickel for the cathodic electrode and nickel-chrome for the anodic one. Plates are built from the metal powders treated by tape casting and sintering processes. Material characteristics such as porosity and density may be properly designed by varying powder particle size, organic binder composition and sintering conditions. These materials may be potentially valuable as acoustic absorbers and their acoustical properties may be adjusted by changing the material's characteristics. Generally materials for MCFC require no impurities and their cost is a critical issue; in acoustical applications, purity requirements are not necessary and scrap samples may be used, contributing strongly to cut costs. This paper deals with an investigation on a typical MCFC material to verify their suitability for noise insulation and absorption. For this purpose, a measurement campaign by Kundt tube on MCFC electrodes was carried out by varying the plate porosity, width, mixture content and purity; optimal configuration has been found in terms of maximum absorption frequency. Results suggested that, for acoustical application, some metal powder maybe substituted by a polymer powder

in order to further reduce costs and further improve absorption. Manufacturing procedures are actually under study.

Materials for Noise Control:

Paper ICA2016-323

Acoustic design of the air transparent soundproofing wall

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Abstract

If air can pass through a noise reducing wall, there are many benefits. Enclosures can reduce heat loads and noise barriers can decrease wind loads. The air transparent soundproofing wall is suggested using plenum chamber arrays. The sound transmission loss can be influenced by several geometric parameters. Among them, there are two key parameters to characterize the acoustic performance. The one is the higher order mode cut-on frequency inside a chamber. The other is the spring-mass-spring resonance frequency. To optimize sound insulation of plenum chamber arrays, two above frequencies need to be designed. The sound transmission loss of the air transparent soundproofing wall was designed and tested. To apply industrial fields as noise barriers or enclosures, the single number rating, Rw, is selected as a performance indicator. The measured transmission loss shows Rw-30 dB (the single number rating). Even this wall is thicker than industrial insulation materials, this wall can allow the natural ventilation and reduce the wind load. If target frequency of the noise reducing measure is fixed, the soundproofing wall can be used in many applications

Friday morning, 9 September 2016 — **Cardenal Pironio Auditorium**
09:40 - 10:40
Noise: Sources and Control
NS1 - Aircraft Noise – Aeroacoustics

Aircraft Noise - Aeroacoustics:

Paper ICA2016-110

Civil aircraft noise reduction: Summary of recent research and overview of forthcoming efforts to promote new research within European context

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Abstract

Over the last decades, the noise by civil aircraft has been mitigated by more than 20 EPNdB. This success was made possible thanks to specific research efforts which focused on reducing the aircraft noise at its source, following the guidelines of International Civil Aviation Organization (ICAO) balanced approach for meeting the international regulations on aircraft noise. On another hand, Advisory Council for Aeronautics Research in Europe (ACARE) now targets a further 50% reduction by 2020 and 65% by 2050 of the perceived noise, with respect to a year 2000 reference. The present paper first recalls some of the research efforts that were deployed within a European context over the last decade and a half. Indeed, since 2000, many Research and Development (R&D) works were performed, which first focused on “technological bricks” and were then progressively replaced by larger Integrated Projects or Joint Research Initiatives aiming at maturing still more the Noise Reduction Technologies (NRTs) proposed, e.g. via a technological transfer from research initiatives to demonstrator platforms. Consequently, the European Research on Aircraft noise is now organized through large programs relying on specific demonstrators. New impetus is expected from mid- and long-term initiatives, such as the EREA recently proposed FutureSky program, which aims at deriving a comprehensive R&D approach that would allow the air traffic growth to meet ICAO and ACARE constraints on perceived aircraft noise. If granted, this European project shall allow strengthening

various national efforts, as well as reinforcing specific cooperation with partners from outside the E.U. (e.g. United States, Russia and Japan).

Aircraft Noise - Aeroacoustics:

Paper ICA2016-93

Flow-induced noise regimes of a wall-mounted finite airfoil

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Abstract

Flow interaction with a wall-mounted finite airfoil is a major noise source in a number of practical aerodynamic and hydrodynamic situations including turbomachinery blade and end-wall flows, aircraft wing and body junction flows and ship appendage and hull junction flows. In all of these cases, the flow around the wall-mounted finite airfoil is three-dimensional with boundary layer impingement at the airfoil-wall junction and flow over the tip. An experimental investigation has been conducted in an anechoic wind tunnel to define the noise characteristics of a wall-mounted finite airfoil with a flat ended tip in cross-flow. This paper describes the wall-mounted finite airfoil noise generation mechanisms and how flow over an airfoil can create tonal or broadband noise. Examples of vortex shedding as well as tonal and broadband noise spectra are presented along with aeroacoustic beamforming sound maps which provide information about the airfoil noise generation mechanisms and noise source locations in each regime.

Aircraft Noise - Aeroacoustics:

Paper ICA2016-214

Laboratory study on jet installation noise effects by designing and using a 3D microphone array

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Abstract

This study is providing the improvement of a progressive acoustic measurement technique for the specific investigation of jet/flap interaction noise. The used microphone array technology has been developed to support a further understanding of the installation noise source mechanisms at UHBR aero-engines and for the modelling of jet/flap interaction noise. Experimental investigations have been performed in the Airbus Group Innovations JetLab, using a laboratory model jet and scaled generic wing/flap system. The focus has been set on the noise source strength and localisation in the isolated case, but especially under the aspect of the further application on installation configurations with different flap angle settings. To achieve this objective, a 3D microphone array has been established and applied to a heated dual jet in the laboratory. The final design of this 3D array will be described as well as its application to installation noise. The experimental work has been carried out in the frame of the research project JERONIMO, funded by the European Commission.

Aircraft Noise - Aeroacoustics:
Paper ICA2016-443

A systematic review of semi-empirical acoustic liner models under grazing flow and high SPL

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Abstract
During the past decades several semi-empirical models have been proposed to predict the acoustic impedance of aircraft turbofan liners based on its geometry and operating conditions. Whereas viscous, radiation and backing effects result from analytical solutions, non-linear effects are often based on curve fitting to experimental data. The problem arises when these equations are applied to different geometries and operating conditions than those used in the fitting procedures since poor agreement between the models can be seen in the literature. Much effort has been made to correct terms which are almost negligible especially when non-linear effects i.e. grazing flow and high sound pressure level are the dominant effects. These terms can easily under- or overpredict the impedance as a consequence of simplistic assumptions or overfitting to the available experimental data. A systematic review of semi-empirical models is done to identify which terms are dominating the impedance estimation (where efforts should be concentrated) at specific conditions. The review also compares the experimental techniques adopted in each study to obtain the impedance values used in the curve fitting procedures, which can have a considerable impact over the resulting semi-empirical models.

INVITED
Aircraft Noise - Aeroacoustics:
Paper ICA2016-131

The influence of an adaptive nacelle inlet lip on fan noise propagation

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Abstract
The aeroacoustic performance of an adaptive inlet of a turbo-fan engine is numerically investigated in this paper. The sound intensity and directivity of the fan noise propagation to the far-field, and the sound level at lateral reference points are investigated. The investigation is performed for three Helmholtz numbers, with the influence of the mean flow included, for a single duct mode (- 8,1). The contour was defined by five movable knots at the leading edge of the inlet. The contour had to fulfil two constraints, namely it had to have a constant length and a convex curvature. The process of contour adaptation was performed in two steps. In the first step, two knots on the inner inlet side were moved in order to attain a certain shape, while other knots were kept fixed. In the second step, the rest of the knots were moved in order to fulfill the constraints. A finite element solver for the Helmholtz equation is used in the inner part of the inlet, with a perfectly matched layer boundary condition close to the inlet entrance. The propagation through the outer part of domain is solved by Kirchhoff integral method. The results show the influence of the inlet shape adaptation on the noise intensity level as

well as the directivity of propagation. The maximum peak intensity level of all inlet shapes is increased by increasing the Helmholtz number. This causes the width of intensity distribution to become narrower and shifted towards the symmetry axis of the nacelle. The inlet shape with the most opened nacelle throat has the lowest peak and an intensity distribution shifted towards the symmetry axis, which indicates the influence of the mean flow. Also, the more closed nacelle throat causes a decrease of the effective perceived noise level.

Aircraft Noise - Aeroacoustics:

Paper ICA2016-108

Experimental study of casing treatment impact on ducted counter-rotating fan noise

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Abstract

Fan of modern turbofan is considered as the most important source of tonal and broadband noise. In addition to the well-studied mechanisms of fan noise generation takes place noise generated by interaction of rotor tip vortex with outlet guide vanes. This component contributes essentially to the overall fan noise. Investigated in several experimental works the detailed tip leakage flow pattern in radial clearance area gives an opportunity to describe possible mechanisms of noise generation connected with a rotor tip vortex. For reduction of fan noise caused by tip vortex various techniques are used. One of them uses an acoustic treatment made of porous material. Besides, presence of porous treatment over the fan rotor leads to easing of stall phenomena in boundary layer on the cowling. This experimental study presents another method of fan noise reduction consisting in installation of slot type casing treatment above the rotors of the ducted counter rotating fan. The test program includes seven casing treatment (CT) configurations: four of them were installed over the second rotor, another three ones – over first and second rotors simultaneously. The ducted counter rotating fan model of 22² diameter has been used as an experimental object. Influence of several configurations of casing treatment on fan noise levels in a far sound field has been investigated.

INVITED

Aircraft Noise - Aeroacoustics:

Paper ICA2016-430

Aero-acoustic fan broadband noise: A new parameterization proposal

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Abstract

Due to the new regulation schedule that is been restricting the aircraft noise emission, the aircraft design processes requires even more reliable aircraft noise prediction models. Especially during take-off, the turbo-fan Engine is one of the noise sources most in concern. Historically the tonal noise have been studied and treated, but the Broadband noise is becoming the aim of studies nowaday. Taking advantage of experimental data that has been collected with the new Aero- Acoustic Fan Rig at University of São Paulo and data of Fan Noise test bed from Aero-Acoustic Noise and Propulsion Lab at NASA Glenn, using the in-duct modal decomposition of the noise, a new parameterization for Broadband noise for turbo-fan are proposed to represent the modal noise spectrum. The new parameteerization are compared with the Gaussian function proposed by Heidmann and even combined together. The results show that the new parametrization represents better the Broadband noise on low frequencies and is good as the Gaussian approach at high frequencies. The combinations of the new function and Gaussian results in an adjustment that the Gaussian function have low influence on final results. The reconstruction of the total spectrum propagated thought the duct shows good agreement with far-field measurements characteristics.

Aircraft Noise - Aeroacoustics:
Paper ICA2016-191
Parsimonious approaches for the laboratory synthesis of wall-pressure excitations
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Abstract
The experimental synthesis of acoustic or aerodynamic wall-pressure excitations over industrial structures, for instance an aircraft sidewall or a car window, is a cost-efficient approach in order to optimise the insulating performance of these structures under real-life forcing generated in a controlled environment. Several strategies like inverse filtering in the spatial or wavenumber domains, holophonic or holographic reproductions, have proved to be successful for the laboratory synthesis of an acoustic diffuse field. But they required a prohibitive number of control sources for the real-time synthesis of sub-wavelength scales such as those associated with the wall-pressures induced by a Turbulent Boundary Layer (TBL). The present study compares the performance of several strategies for a direct reproduction of TBL excitations, independently of the test panel physical properties and with a reasonable number of sources. A first approach, the focused synthesis, uses spatial windowing to reduce the surface over which the TBL is reproduced, thereby enlarging the range of synthesized wavenumbers in the subsonic domain, if possible beyond the convective ridge. The two other approaches aim at synthesizing a reduced-rank approximation to either the target TBL correlation function or the source-fields acoustical transfers. The efficiency of these methods will be examined in terms of both the reproduction accuracy and the required number of sources that can be used to achieve the synthesis over a broad frequency range.

Friday morning, 9 September 2016 **Room 204**
09:20 - 10:40
Electroacoustics and Audio Engineering
EL1 - Electroacoustics and Audio Engineering

INVITED
Electroacoustics and Audio Engineering:
Paper ICA2016-290
Frequency distribution of temporal sound pressure capacity requirement in two-way and three-way active monitors
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Abstract
Active monitoring loudspeakers are optimized for very high sound pressure output using very high power amplifiers in an enclosure having a compact physical size. All drivers in such a system use thermal overload protection to limit heating in the voice coils. Optimal protection allows full lengths of the typical audio events to be reproduced before the thermal protection must activate. This work studies the temporal statistics of an audio signal to determine how the limit to the duration of the maximum power level output must be chosen for minimum audible impact. The statistical distribution of the audio event durations in cinematic 24 bit word length, 48 kHz sample rate audio track and 16 bit word length 44.1 kHz stereo audio tracks are studied for the two-way and three-way active monitoring systems to determine the temporal capacity requirement as a function of frequency. This data is analysed for each driver output channel separately in the case of a two-way monitor and a three-way monitor. Statistics are presented to describe the expected maximum sound pressure level audio event duration as a function of frequency. The mean maximum level acoustic event duration is inversely proportional to the low corner frequency of the driver channel bandwidth. Music sound tracks and film sound tracks do not differ in this respect even if the music sound track crest factor can be significantly smaller.

INVITED

Electroacoustics and Audio Engineering:

[Paper ICA2016-436]

Loudspeaker protection system for mobile devices

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Abstract

For mobile devices sound becomes more and more important. However, due to the miniaturization of mobile communication devices (smart phones, tablets, smart watches etc.) the demands for acoustic output are very difficult to achieve. The size of the transducer becomes smaller and smaller and the driving power very often reaches the limit of the transducers capability even at very moderate loudness. It is therefore very likely to either damage the loudspeaker or to create an unwanted amount of distortion. The proposed method describes a loudspeaker control unit that allows both equalization of the output with respect to the momentary state (heat, excursion etc.), limitation of the input power with respect to mechanical (excursion) and thermal limits of the transducer. In dependence of the temperature several parameters (DC resistance and compliance) are predicted so to correct for the effects in the TS-parameters connected to this. The final algorithms show that with this approach a better reproduction quality in connection with higher output can be achieved, though the risk of damaging the system has been reduced to a minimum. Besides the system description results with running speech and music will be presented.

INVITED

Electroacoustics and Audio Engineering:

[Paper ICA2016-655]

Measurement of the contribution to the acoustical impedance of a loudspeaker due the internal cavities in magnetic circuit using an impedance tube

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Abstract

The acoustical back loading due to the inner cavities of the magnetic circuit can influence in the frequency response of a loudspeaker, therefore in order to optimize the design; it is useful to know the behavior of this loading. In this paper, it is shown that it is possible to measure the acoustical impedance of single elements or combination of these, e.g. resonators located inside the magnetic circuit using an impedance tube. The method presented here is meant to be used in the development stage of the loudspeaker when it is still possible to remove the voice coil or to close partially or totally some of the inner cavities.

Electroacoustics and Audio Encineering:

[Paper ICA2016-355]

Two modified IEC 60318-4 ear simulators for extended dynamic range

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Abstract

The international standard IEC 60318-4 specifies an occluded ear simulator for testing headphones, earphones, hearing protectors, hearing aids etc. The standard specifies a specific microphone type which limits the dynamic range of the ear simulator, such that it is not possible to measure very low levels or very high levels. Additionally, the standard 711 ear simulator is often interfaced to a pinna simulator incorporated in a Head and Torso simulator as per IEC 60318-7. This interface has traditionally been implemented as a cylindrical, straight ear canal simulator. This makes the fit of many modern in-ear headphones and hearing protectors problematic and unrealistic. By using low noise microphones instead of the standard microphones, the ear simulator can be used for measuring extremely low sound pressure levels such as noise floor, low level distortion or microphonics. Conversely, by using low sensitivity microphones, the ear simulator can be used for extremely high level measurements—useful for testing active and passive attenuation ratings of hearing protectors. Moreover, using a vast database of 3D human ear canal scans, a new pinna and ear canal simulator is proposed that will greatly improve measurement accuracy and repeatability on products going on or in the ear.

Friday morning, 9 September 2016 **Room 204**

11:00 - 13:00

Electroacoustics and Audio Engineering

EL1 - Electroacoustics and Audio Engineering

Electroacoustics and Audio Engineering:

[Paper ICA2016-810]

Effective radiation area (Sd) for an axisymmetric piston radiating in an infinite baffle

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Abstract

Exact calculation and experimental determination of the Effective Radiation Area (SD) has always been a very important topic in order to establish loudspeaker parameters. Even if the Rayleigh Diffraction Integral can be used to evaluate sound pressure at any point, is very difficult to obtain a precise value for the SD because of the solution of the surface integral. This paper, propose an easier method using computational tools and the cylindrical symmetry in most of the pistons, in order to reduce the mathematical difficulties. Then, the obtained solutions for different kind of piston are compared with other solutions obtained in different experimental approaches in order to determine the accuracy of this new approach.

Electroacoustics and Audio Engineering:
Paper ICA2016-215
Implementation of a low cost remote failure monitoring system for speaker lines in a shopping mall
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Abstract
Nowadays, Public Address (PA) designs are primarily used for background music and evacuation messages broadcast during emergencies. For the latter, it is very important to detect if there is a partial failure, disconnection or short circuit in the speaker lines. In this work, a very low cost remote failure monitoring system for speaker lines is presented in detail. A relay array, controlled by a Raspberry Pi computer, is in charge of switching between the different lines. Each line is measured with test signals between 80 Hz and 10 kHz. Finally, the information is sent to a cloud-database. The final user can access to it through a web interface, and receive alerts if a failure is detected. The results from a shopping mall's PA system monitoring are presented.

Electroacoustics and Audio Engineering:
Paper ICA2016-155
Development of multichannel single-unit microphone using shotgun microphone array
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Abstract
We developed a multichannel single-unit microphone using circular shotgun microphone array for simple recording of multichannel audio, such as 22.2ch audio. The microphone array consists of 8 shotgun microphone elements arranged at 45 degree intervals. Though sharp directivity corresponding to the direction of each element is preferable, directivity at low frequency is wide (directivity of a shotgun microphone becomes sharp with increasing frequency due to the acoustic tube). To improve directivity at low frequency, directivity control by digital signal processing can be applied. We evaluated the performance of our multichannel single-unit microphone through numerical simulation and experiment.

Electroacoustics and Audio Engineering:
Paper ICA2016-193
Modeling of the electroacoustic coupling of electrostatic microphones including the preamplifier circuit
Bernardo Henrique Pereira Murta[(a)], **Eric Brandão**[(b)], **Julio Cordioli**[(c)],
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Abstract
This research aims to study tools to model and design electrostatic microphones coupled with its preamplifier circuits. The outcome is the access to their combined sensitivities curves, which allows the design of microphones with a wider and flat bandwidth. Analytical and numerical modeling techniques are explored and compared. On one hand, the lumped parameters approach is the basis of the analytical modeling of acoustic transducers. That is, this technique allows the engineer to design the transducer and its preamplifier circuit by predicting its sensitivity changes due to variations of model properties with low computational cost. On the other hand, numerical analysis is carried out using the Finite Element Method with a multiphysics approach, which is able to solve both the transducer model

and the coupled electrical circuit. Two microphones with different complexities and constructive characteristics are studied. For validation of the proposed techniques, the behavior of a commercial measurement microphone model that has been well studied in the literature is considered. Once the validation of the modeling approach is satisfactory, one can use the same methodology to study a piezoelectric microphone for hearing aid applications, for instance. Its frequency response requires a designed preamplifier which should be able to make its sensitivity flatter over the audio bandwidth and to improve its output voltage level. The research also objectifies to analyze the whole chain of energy transducing and signal conditioning in order to prepare the fully coupled model for optimization procedures. The goal is to conceive efficient high-performance systems with low cost hardware.

Electroacoustics and Audio Engineering:

Paper ICA2016-147

An investigation about diffuse-field calibration of measurement microphones by the reciprocity technique

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Abstract

In the calibration of measurement microphones by the reciprocity technique, microphone sensitivity is usually determined from the electrical transfer impedance and the acoustical transfer impedance between three microphones acoustically coupled in pair-wise combinations. This calibration is well known in pressure-field and in free-field conditions but it is under research for diffuse-field. In this paper is presented a proposal to perform this calibration in diffuse-field. The microphones are placed in a small reverberation chamber with boundary (volume) diffusers. The electrical transfer impedance is obtained from the average of measurements at different positions in the chamber. In each measurement, the reverberation is separated from the direct sound using a suitable window function. The acoustical transfer impedance is obtained from the chamber reverberation time, which is determined using the same measurements employed to obtain the electrical transfer impedance. The results support the viability of the proposal.

Electroacoustics and Audio Engineering:

Paper ICA2016-47

Analysis on the interaural level difference in near-field-compensated higher order Ambisonics reproduction

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Abstract

Near-field-compensated higher order Ambisonics (NFC-HOA) is a spatial sound reproduction technique based on spherical harmonics decomposition and each order approximation of sound field. The aim of NFC-HOA is to reconstruct the curve wavefront of spherical wave cause by sound source at different distances. It is desired that NFC-HOA is capable of recreating appropriate interaural level difference (ILD) which is considered to be an auditory distance localization cue for nearby sound source within a distance of 1 m relative to head center and outside the median plane. The present work analyzes the ILD in NFC-HOA reproduction by using head-related transfer functions and compares with the case of a real point source. The results indicate that, due to the requirement of excessive low-frequency boost in the distance-compensated filters, it is difficult for NFC-HOA to recreate appropriate ILD for nearby target virtual source below the frequency of 0.7 to 1kHz. On the other hand, in order to recreate appropriate high-frequency ILD for nearby target virtual source, a much higher order NFC-HOA is needed. An illustrative example indicates that, even for the center listening position, a 12-order NFC-HOA with not less than 169 loudspeakers is needed for recreating appropriate ILD of a lateral virtual source at 0.25m and up to the frequency of 5kHz. Therefore, in practice, NFC-HOA is workable in certain mid-frequency range.

Numerical Computation in Musical Acoustics:
Paper ICA2016-54

Chaotic behavior of the piccolo

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Abstract
A direct numerical solution of the Navier-Stokes equations in three dimensions has been used to compute the sound pressure produced by a piccolo as a function of time *p(t).* For moderate blowing speeds *u*, a pure tone is produced, but as *u* is increased *p(t)* exhibits an increasingly complex behavior. The behavior of *p(t)* is consistent with a positive Lyapunov exponent at high values of *u*. Detailed results for the power spectrum reveal a simple pure tone dominated by a single frequency at low *u*, as expected. As *u* is increased additional frequencies appear in the spectrum along with broadband noise in certain spectral regions. The results suggest that the piccolo is, under certain blowing conditions, a chaotic system.

Numerical Computation in Musical Acoustics:
Paper ICA2016-399

Characterisation of brass instruments with mutesthrough experimental means and finite-element simulations

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Abstract
A number of mutes have traditionally been used to modify the timbre and volume of the sound of brass instruments. In the present work, the effects of adding straight, cup, and Harmon mutes to trumpets and trombones are analysed and discussed initially through the measurement of input acoustic impedance and frequency response. The input impedance was obtained in a frequency range between 50 and 3000 Hz, and the frequency response was measured between 50 and 6400 Hz. These results were then compared with finite-element simulations of acoustic propagation in the complete instrument with and without mutes. These comparisons allow for validation of the numerical method, and make numerical experimentation possible with mutes with a wide variety of geometries, potentially helping to save time and reduce costs during the design process. X-ray images of the mutes and instruments were used to define their contour by means of a CT Nikon Metrology XT H225ST system.

Numerical Computation in Musical Acoustics:
Paper ICA2016-459

A numerical model for axial strain vibrations of brass wind instrument bells offering superior performance compared to finite element methods

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Abstract

Experimental observations suggesting an effect of wall vibrations on the radiated sound of wind instruments have been published by many different authors over a very long period of time. Simultaneously there was an ongoing debate about possible physical mechanisms behind that phenomenon. However, in the case of brass wind instruments a theory which is consistent with experimental results has been developed rather recently. It suggests that axial strain vibrations of the bell are responsible for the observed interaction between structural vibrations and the radiated sound. In this paper a numerical model of such axial vibrations in brass wind instrument bells is proposed, which is computationally much less expensive than Finite Element Method (FEM) simulations but comparably accurate. It takes advantage of the relatively simple axisymmetric and thin-walled structure of bells and the knowledge that several degrees of freedom can be neglected in that particular problem. The computational efficiency is crucial to facilitate the analysis of coupling between structural and acoustic vibrations in real time or during a design optimization process. The proposed model is based on a frequency domain approach published by Kausel et al. in JASA 137(6) where the bell's mass and stiffness is represented by a two-dimensional network of lumped masses and springs. Here it is extended for increased accuracy in steeply flaring sections of the bell. This is achieved by taking bending stiffness into account in the region near the rim where structural stiffness is no longer the dominant restoring force.

Numerical Computation in Musical Acoustics:
Paper ICA2016-560

Modelling collisions of nonlinear strings against rigid barriers: Conservative finite difference schemes with application to sound synthesis

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Abstract

Strings are common elements found in many musical instruments. Various models of string dynamics exist, describing cases of increasing complexity. For fine-grained simulation of string dynamics, either in the context of musical acoustics investigation or for sound synthesis, linear models such as the wave equation with stiffness are, however, insufficient. Recent work has focused on the coupling of a Kirchhoff- Carrier nonlinear string model with collisions against lumped or distributed barriers, showing promising results. The collisions are described by means of a penalty potential, relying on a fictitious interpenetration but allowing a description within an energy-balanced framework. In this work, the same collision model is used, but the nonlinear string model is further developed, in order to allow complex modal coupling rules, as well as amplitude-dependent pitch. In order to handle such complex system, appropriate finite difference schemes are developed, using energy-balanced methods. Results of simulations are presented, along with some applications to sound synthesis.

Wind Instruments:
[Paper ICA2016-748]

Reed chamber resonances and attack transients in free reed instruments

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Abstract
Western free reed instruments such as the accordion, harmonica, and harmonium do not normally employ pipe resonators to determine the pitch, but all do feature some sort of reed chamber or cavity in which the reed is mounted. The reed chamber will necessarily have resonances which can affect the tone quality and may have some effect on the pitch, but, since the cavity volumes are small and the resonances have high frequencies, the effects on the reed vibration tend to be small. An exception to this can occur in the accordion or harmonica for higher pitched reeds, for which a resonance of the reed chamber can be close to the vibration frequency of the reed tongue. In this case the cavity air vibration can possibly interfere with tongue vibration, inhibiting the sounding of the reed. For various configurations of the reed chamber, reed motion during the initial transient stage of vibration has been analyzed, exploring the role of transverse and torsional modes in the early stages of the transient, as well as effects on the rise time and final amplitude of vibration due to unfavorable reed chamber configurations. [Work partially supported by United States National Science Foundation Grant PHY-1004860]

INVITED
Wind Instruments:
[Paper ICA2016-756]

Aeroacoustics of free reeds

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Abstract
Free reeds, such as those found in the accordion and the harmonica, produce sound through complex flow-structure interaction. This study uses extensive experimental measurements of acoustic, aerodynamic and vibration phenomena to develop an improved physical understanding of how a free reed produces sound. We propose a new model for the instability of the reed and for how the oscillation of the reed tongue generates sound, examining how the characteristics of the sound change with the key parameters. Laser vibrometer and high speed camera measurements were used to examine the motion of free reeds. To characterise and distinguish the aeroacoustic sound sources, directivity measurements with far-field microphones were carried out, along with an inspection of the acoustic waveform's causal relationship to the position of the reed in its cycle. The experimental data matches well with simple theoretical modelling of the aeroacoustic sources. The key sources of sound were identified to be a dipole source due to the fluctuating force exerted on the fluid by the moving reed tongue, and a monopole source associated with the fluctuating mass flow of air through the reed slot. We show that the mass flow fluctuation is the dominant mechanism of sound radiation from free reeds.

INVITED

Wind Instruments:

Paper ICA2016-610

Validation of brass wind instrument radiation models in relation to their physical accuracy using an optical schlieren imaging setup

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Abstract

The complexity of brass instrument radiation theory has led to a large number of hypotheses and simplifications, which allow for the development of computable models. Most of these, used in the calculation of input impedances, sound simulation and instrument auralisation amongst others, are yet to be validated experimentally. Schlieren imaging techniques permit visualisation of pronounced gradients in the refractive index of air, caused in turn by sharp changes in the density of air, such as the ones which occur during non-linear sound propagation in some loudly played brass wind instruments. In this study, a Schlieren optical setup is used to visualise the shape of the wave fronts radiated by a group of brass instruments with different degrees of flaring in their bells. The geometry of these wave fronts, and its evolution after leaving the instrument, is characterised and related to the flaring parameters for each instrument, providing grounds for evaluating the physical accuracy of existing models.

Wind Instruments:

Paper ICA2016-347

Influence of strain-gauge sensors on the vibrational behaviour of single reeds

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Abstract

Experimental measurements are often used in conjunction with physical modelling to characterise sound generation in musical instruments. Focusing on single-reed woodwind instruments, such analyses have provided accurate descriptions of the coupling between the sound excitation mechanism and the resonator during steady-state regimes. For note transients however, more detailed measurements of the reed vibrations under real playing conditions are required. Therefore, strain gauge sensors haven been placed on a series of clarinet and saxophone reeds, in order to capture the vibrations without interfering with the player. Different ways of attaching the sensors to the reeds are considered and the resulting influence is quantified by means of Laser Doppler Vibrometry and Electronic Speckle Pattern Interferometry.

Wind Instruments:
[Paper ICA2016-870]

Effect of input signal shape on the nonlinear steepening of transient acoustic waves in a cylindrical tube

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Abstract
Nonlinear acoustic propagation effects are known to account for waveform steepening for sufficiently intense signals having travelled over a long enough distance. This steepening, which will eventually produce a shock wave, results, in turn, in a transfer of energy to the high end of the frequency spectrum. The shock formation distance, however, is inversely proportional not to the maximum amplitude of the waveform, but to its maximum slope. We test this theoretical result by producing two sets of short pulses in a long cylindrical tube, where the energy content of each pulse in a set is constant, but the maximum slope varies. The pulses are allowed to propagate over a distance long enough for nonlinear steepening to become apparent. We observe the expected result, where for an initially loud signal the value of the maximum slope does affect the rate at which energy is pumped to high frequencies, whereas for a signal with much smaller initial energy content it does not. This result is of interest in the context of musical acoustics, as it confirms recent observations where players of some brass instruments can affect the "brassiness" of their sound purely through slight changes in embochure.

Friday morning, 9 September 2016 **Auditorium 2**
09:20 - 10:40
Noise: Sources and Control
NS5 - Sustainable Materials for Sound Absorption and Insulation

INVITED
Sustainable Materials for Sound Absorption and Insulation:
[Paper ICA2016-605]

A micro-perforated panel absorber with periodic sub-cavities at different depths by quadratic residue sequences

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Abstract
A micro-perforated panel (MPP) absorber supposed to have super-broad sound absorption frequency band is presented in this paper. This kind of MPP absorber was designed by covering a MPP layer over periodic sub-cavities at different depths by quadratic residue sequences. In this paper, firstly, an analytical model is proposed to predict the normal incident sound absorption performance of this MPP absorber in its design stage. Secondly, finite element procedure and experimental measurements were employed and conducted to provide validation on the proposed analytical model and on the sound absorption performance of this MPP absorber. Results show that, the analytical model provides simple yet accurate prediction on the sound absorption performance of this MPP absorber. It is also shown that, the MPP absorber can have normal incidence absorption coefficients higher than 0.5 over the frequencies from 450Hz to 3500Hz.

INVITED

Sustainable Materials for Sound Absorption and Insulation:

Paper ICA2016-188

Micro-Perforated materials for the reduction of flow-induced noise

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Abstract

Turbulent Boundary Layer (TBL) induced noise is one of the dominant noise sources in modern aircraft, where classical materials are progressively being substituted by fiber-reinforced polymers. Although passive methods continue to be developed due to their straight realization in practical flight applications, they add additional weight and are also accompanied by some degradation in high-lift performance. We propose in this work to study the behaviour of insulating partitions composed of Micro-Perforated Panels (MPPs) subject to a TBL excitation, for replacing or complementing passive or active flow noise control solutions. We have carried out a set of simulations to establish a comparison between the performance of the MPP control devices when varying the physical configurations of the partition and the nature of the primary noise excitation. It can be shown that when exciting the partition with less correlated random loads, the corresponding TL increases progressively. For the model validation, experiments have been performed in the low-speed wind tunnel of the IRPHE Fluid Dynamics Laboratory in order to determine the acoustic and aerodynamic TL performance of a number of MPP multilayer partitions (double and triple layers) when subject to a TBL of free stream velocity 30.7 m/s. The effect of inserting a micro-perforated panel within the cavity at unequal distances from the front and back panels dampens more efficiently the Mass-Air-Mass controlled resonances of the Panel-Cavity-Panel system with respect to those of the MPP-Cavity-Panel system, already damped by the front MPP. This results in a higher TL difference between the triple and double partitions with a plain front panel, which is about 8 dB, with respect to the same configurations with a microperforated front panel, which is about 4 dB.

Sustainable Materials for Sound Absorption and Insulation:

Paper ICA2016-852

Acoustic absorbers based on recycled materials

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Abstract

Acoustic absorbers are usually expensive materials, which makes them difficult to afford for third-world countries educational premises. This is one of main the reasons why classroom acoustics are frequently poor, affecting educational quality. To circumvent that problem, a material based on recycled paper, rice husks and additives has been developed. Emphasis has been made on getting a handcraft manufacturing process capable of being carried out by the school community members. Another important condition that was proposed is that the manufacturing process uses the minimum amount of water and that it uses a solar drying method. Preliminary results are reported..

Sustainable Materials for Sound Absorption and Insulation:
[Paper ICA2016-201]

Sound insulation research of frameless three-layers concrete partitions

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Abstract
Recently the frame method of construction of multi-story buildings has been widely developed. In some cases, primarily in Russia, the use of frame walls between flats, which insure required sound insulations, is extremely limited by mentality of residents, who perceive their flats by the principle «my house is my fortress». Due to the limited load on the floor, structures lightweight gypsum or aerated concrete blocks are used for this purpose, but they do not have sufficient soundproofing. Therefore, it is necessary to provide the required sound insulation of walls between flats, which should be made of concrete and have significantly reduced weight and acceptable thickness. For this purpose three-layer frameless barriers were developed. Outer layers of these barriers are made of concrete plates with thickness of 30 mm ... 50 mm. Total thickness of the barriers do not exceed 160 mm ... 200 mm. These barriers provide sound insulation of 52 dB ... 53 dB. The stability of the plates is provided by their form. Below are given the results of research of how thicknesses of the plates and their inner layer, material of the inner layer and density of the concrete plates influence sound insulation.

Friday morning, 9 September 2016 **Auditorium 2**
11:20 - 13:00
Psychological and Physiological Acoustics
PP1 - Free-Field Virtual Psychoacoustics and Hearing Impairment

INVITED
Free-Field Virtual Psychoacoustics and Hearing Impairment:
[Paper ICA2016-711]

Hearing impairment, hearing aids, and cues for self motion

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Abstract
When listeners turn their heads, the resulting change in binaural level and timing cues constitutes useable information about the location of signals in the world, particularly on front/back location. However, in order to make use of these dynamically changing cues, listeners must be able to compare the speed and direction in which a signal moves with the speed and direction of their head movements. Hearing impairment is frequently co-morbid with vestibular impairment, rendering access to self-motion cues less reliable, and it is also associated with an increase in the minimum audible movement angle, a measure of auditory motion processing. Hearing impairment is also typically associated with raised thresholds in the range of frequencies in which the filtering effects of the outer ear provide other cues useful for resolving front/back localization ambiguities. By moving sound sources as a function of the instantaneous position of the listener's head, we created a front/back illusion, and demonstrated that listeners with hearing impairment rely more heavily on self-motion cues than on high frequency information, even though their ability to use binaural cues can be impaired. The use of hearing aids had heterogeneous effects in different listeners, although in no case did they return a listener to normal performance. To examine this phenomenon we made recordings of the output of hearing aids driven with a structured noise sequence whose level transitions were statistically controlled. We found that hearing aids affect spectral cues as well as interaural level and temporal envelope differences. Taken together with recordings made on a rotating KEMAR manikin, we demonstrate that hearing aids may interfere with a listener's ability to process acoustical cues for self motion.

INVITED

Free-Field Virtual Psychoacoustics and Hearing Impairment:

Paper ICA2016-653

Factors influencing the ecological validity of laboratory-based speech tests

J. M. Buchholz [a, b], A. Westermann[a], A. Weisser[a], T. Beechey[a], C. Oreinos[a], V. Best[c], G. Keidser[a]

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Abstract

Laboratory-based performance measures of speech communication ability and hearing device benefit often do not correlate well with the performance reported and experienced by hearing-impaired subjects in the real world. The main reasons are the unrealistic stimuli as well as the speech tasks that are commonly applied. This study first provides an overview of the acoustic factors that need to be addressed to improve the ecological validity of laboratory based speech tests. A number of possible solutions and their practical limitations are then discussed and verified with experimental data. Loudspeaker-based sound reproduction is used to create realistic acoustic environments and their important properties are evaluated using acoustic measures as well as measures of informational masking. A speech conversation task is applied to derive both realistic signal-to-noise ratios and to evaluate the impact of the Lombard effect on speech outcomes. Finally, additional factors are discussed, including the impact of the impaired auditory system (e.g., reduced audibility), visual cues, and using a speech comprehension task instead of a more common sentence recall task.

INVITED

Free-Field Virtual Psychoacoustics and Hearing Impairment:

Paper ICA2016-431

Speech perception by children in a real-time virtual acoustic environment with simulated hearing aids and room acoustics

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Abstract

Classrooms with demanding acoustic requirements for children fitted with hearing aids can be simulated effectively by real-time virtual acoustic environments. Physical accuracy is achieved using room impulse responses and a binaural reproduction system extended by research hearing aids. The generation of virtual sound sources is based on individualized head-related and hearing aid-related transfer functions. For the simulation of hearing aid algorithms, a software platform, which utilizes individual audiograms to generate fitting curves, processes the signals before being reproduced. In this study, a release from masking paradigm by Cameron and Dillon (2007) was adapted to assess speech intelligibility by children fitted with hearing aids in realistic reverberant environments. Speech reception thresholds are measured in the realistic acoustic scenes with room acoustics and compared to results from age-matched normal-hearing children.

Free-Field Virtual Psychoacoustics and Hearing Impairment:
Paper ICA2016-776

Auditory localization of real and virtual sounds by hearing impaired listeners

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Abstract
Many studies have evaluated the performance of virtual audio displays with NH listeners, but very little information is available on the effect that hearing loss has on the localization of virtual sounds. In this study, normal hearing (NH) and hearing impaired (HI) listeners were asked to localize sounds of short (250 ms), medium (1000 ms), and long (4000 ms) duration both in the free field and with a head-tracked virtual audio display. Head-Related Transfer Functions (HRTFs) measured on a KEMAR manikin were used in the virtual display to explore whether HI listeners with high frequency hearing loss might be less susceptible to the distorted pinnae cues present in non-individualized HRTFs than NH listeners. The results show that the HI listeners localized sounds less accurately than the NH listeners, and that both groups consistently localized virtual sounds less accurately than free-field sounds. They also showed that both hearing impairment and the use of non-individualized HRTFs tended to result in a systematic upward bias in response elevation. Contrary to our initial hypothesis, these results suggest that the high-frequency distortions introduced by non-individualized HRTFs are effectively additive with the high-frequency distortions in localization cues introduced by hearing impairment. The results did, however, show a high correlation between free-field and virtual localization performance in the HI listeners, suggesting that virtual audio display systems who have some utility as a clinical tool to identify individuals who have much worse than normal localization performance in the free field.

Free-Field Virtual Psychoacoustics and Hearing Impairment:
Paper ICA2016-123

Study on evaluation of speech intelligibility focusing on speech privacy

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Abstract
Recently, concerns about speech privacy are increasing continuously in Japan. In order to achieve the speech privacy, proper evaluation method of the speech privacy is required. Speech intelligibility has a high relationship with the speech privacy. However, the relationship between impression of oral information leakage and the speech intelligibility is not clear in Japan. In this study, the speech intelligibility test and categorical rating tests for investigating the impression of oral information leakage were measured according to the change in the speech levels. In addition, influences of background noise on the speech intelligibility test and the categorical rating tests were also examined.

Architectural Acoustics for Non-Performance Spaces:

[Paper ICA2016-713]

Transparent micro-perforated sound absorbers revisited

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Abstract

More than 15 years after the first applications of micro-perforated sound absorbers in architectural acoustics there still is a growing demand of this kind of sound absorbers. Based on the theory of micro-perforated panel sound-absorbing constructions by D. Y. Maa in 1975 various materials have been used as micro-perforated sound absorbers. Fully transparent sound absorbers as well as printed and translucent materials allow a combination of acoustic and light design. 3D-shapes used as lamps and other applications have become available. Measurements for different set-ups will be presented as well as applications in various projects will be discussed. Metal, wood, polycarbonate plates and foils as well as other sheets have been micro-perforated. In this contribution a short review of the applications of various different materials with transparent micro-perforated sound absorbers will be presented.

Architectural Acoustics for Non-Performance Spaces:

[Paper ICA2016-383]

Effects of absorption and scattering coefficient uncertainties on levels and reverberation times computed with soundparticle simulation: A case study

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Abstract

Room acoustic simulations depend on input parameters, especially those describing acoustic properties of a room's surface. Surface parameters usually are absorption and scattering coefficients. Round robin comparisons on the same specimen in different laboratories show deviations and thus uncertainties, even if conducted according to standards. Moreover, there is little data on scattering coefficients available, thus reasonable guessing is a common practice, yielding even larger uncertainties. In particular, reverberation times can depend drastically on the scattering coefficients. In this paper, effects of both absorption and scattering coefficient uncertainties on sound particle simulation method (SPSM) results are investigated. Sound Intensity level and reverberation times are examined. This is done with a systematic set of case studies: several room proportions and shapes (mainly but not only rectangular), combinations of both parameter values as well as their spatial configurations are examined. In future work, these results can be used to either specify the needed accuracy of surface parameters or to give an upper limit for a reasonable SPSM accuracy. They may also help to distinguish types of rooms where Sabine's theory can be applied from those where it fails.

Architectural Acoustics for Non-Performance Spaces:
[Paper ICA2016-464]

Efficient prediction of low-frequency sound fields in rectangular rooms with localized boundary absorption

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Abstract
The acoustical behavior of non-performance spaces is usually described by a single quantity, the space-averaged reverberation time, which in predictions is obtained from the room volume and the total amount of absorption using Sabine's formula. While this descriptor is sufficient for predicting the listening experience when the sound field in the room is effectively diffuse, this is typically not the case in small rooms and at low frequencies: in such cases, the effect of location (corner or surface), variation in distribution (homogeneous or patchwork) and size (boundary effects) of sound absorbing materials on the overall sound absorption can be important. Even in standardized laboratory conditions for measuring sound absorption based on the Sabine formula, significant differences between measurement results from different laboratories have been observed for similar materials, especially at low frequencies. In order to gain insight into these size, distribution and location effects, an efficient method is proposed for predicting the sound fields in rectangular rooms with localized boundary absorption. The absorption of the finishing materials is accounted for by means of a frequency-dependent acoustic impedance. A Lagrange-Rayleigh- Ritz approach is followed, in which the acoustic modes of the equivalent hard-walled room are used as basis functions, to which specific global basis functions are added that capture the finite impedance boundary conditions.

Friday morning, 9 September 2016 **Auditorium 3**
11:00 - 13:00
Architectural Acoustics - Room and Building Acoustics
AA3 - Architectural Acoustics for Non-Performance Spaces

Architectural Acoustics for Non-Performance Spaces:
[Paper ICA2016-156]

Room acoustical optimization: Average values of room acoustical parameters as a function of room shape, absorption and scattering

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Abstract
Numerical methods like ray tracing nowadays allow a comfortable computation of room acoustical parameters like reverberation time (RT), definition (D), lateral efficiency (LE) and strength (G). However, they do not deliver rules for an optimum room design. It is an old dream to have an inverse method that, for some given target parameters describing room acoustical qualities, delivers the (or one) optimum room shape and the distribution of absorbers and diffusers. Going only a small part of that way, this approach aims at to just estimate average values of some room acoustical parameters as a function of room volume and proportions, mean absorption values, and source-listener distance. By methods of geometric-statistical room acoustics, some relations are derived from average echograms and average time delays of first reflections and verified by ray tracing experiments. Concerning the important lateral reflections, non-diffuse sound fields have to be considered, so numerical methods are needed, however, may be restricted to 2D. The relationships between different shapes of ground plans (like rectangular, trapezoidal or circular, or different kinds of zick-zack side walls), scattering coefficients and lateral efficiency are investigated by sound particle simulations.

Architectural Acoustics for Non-Performance Spaces:
Paper ICA2016-127

Efficiency factors characterizing sound reflection properties of a room ceiling

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Abstract
The shape of a room ceiling influences acoustics of the room dramatically. Bad shape causes many acoustic defects like sound focusing, high delays, lack of reflections, whereas good shape provides useful support of the direct sound and as a result increases loudness, improves speech and music intelligibility. Many books give recommendations on the ceiling geometry, which has to be designed to reflect sound to the rear of the room, or to diffuse it throughout the room. But any quantitative parameters for characterizing efficiency of sound reflection from the ceiling are not applied in practice. It seems that it would be useful to specify the room ceiling by any parameters describing its reflection properties. For this purpose two efficiency factors in framework of geometrical acoustics are proposed in presented paper. First factor is a ratio of the sound energy reflected from the ceiling towards seats to the sound energy incident to the ceiling. At the best, this ratio equals 1, but not always maximal ratio corresponds with good ceiling shape. For this reason second factor is also used, it is a ratio of the sound energy reflected from the ceiling towards seats to sound energy radiated by a source into the room. Maximal value of second factor is defined by a ratio of the room height to the room length. The room with a balcony requires refinement of efficiency factors definition. Possible way is to separate the sound energy reflected by the ceiling into the energy reflected towards the main floor and the energy reflected towards the balcony. Proposed efficiency factors are analysed for four classic concert halls with excellent acoustics. It provides preliminary values of the factors suitable for evaluation of acoustic properties of the room ceiling.

Architectural Acoustics for Non-Performance Spaces:
Paper ICA2016-197

Physically-based numerical sound propagation modeling in rooms with non-flat walls

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Abstract
Nowadays, accurate sound propagation modeling is one of the main research axes in room acoustics. Many numerical methods exist, each with their own benefits. Nevertheless, compromises in terms of accuracy, size of the studied domain or frequency range of calculation must be made even for the most advanced models. The objective of this work is to develop a numerical method modeling the sound propagation in an enclosed space (e.g. an industrial workplace) and taking accurately into account the sound scattering influence of the geometrical irregularities of the walls (cavities, beams, windows, etc.). The method developed in this paper is based on the adaptive rectangular decomposition method (ARD) with an improved consideration of the boundary conditions. First, this study describes a way to improve the calculation and to reduce the error induced by perfectly matched layers (PML) used to absorb sound waves at the room boundaries. Then, it details how to implement frequency-dependent reflection at the boundaries using digital impedance filters (DIF) and boundary conditions based on the finite-difference time-domain method (FDTD). Finally, the model was validated, first by comparison with calculations using the Kobayashi Potential method and with measurements, both carried out to obtain the acoustic pressure above a complex surface constituted by rectangular cavities in free field conditions. Then, it was validated by comparison with the image source method (IS) and measurements in a real room.

Architectural Acoustics for Non-Performance Spaces:
Paper ICA2016-900

Study case of a public hospital

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Abstract
An invisible factor is noise, as well as noise control and elimination in the different areas of the hospital. According to various studies conducted in different parts of the world, noise induces an increased risk of medical errors, contributes to staff stressing and burnout, and affects patient length of recovering. According to the World Health Organization (WHO), it interferes in speech perception including the abovementioned disorders. The study is focused on the analysis of sound in a sensitive area of a Public hospital, representing other acoustically implicated equivalent areas, such as corridors and circulation areas, inpatient settings, hospital waiting rooms, neonatology. In situ RT60 Reverberation measurements have been performed in accordance with international standards. Different constructive materials have been used in order to compare results on the acoustic comfort, which proved that hospital noisy areas may be improved, even in those cases where this fact has not been considered in the original project.

Architectural Acoustics for Non-Performance Spaces:
Paper ICA2016-313

Multiplex cinema halls: Design and construction of six halls in the city of Mar del Plata

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Abstract
A shopping promenade that incorporates six cinema halls with capacities ranging from 150 to 310 locations has been built in the city of Mar del Plata, Argentina. The authors of this project were the architects Mariani, Pérez Maraviglia and Cañadas. The employer of the project was Florencio Aldrey Iglesias, who has a long and recognized trajectory both locally and nationally. The place chosen by the company was the former terminal bus station in the city, maintaining the main building, but reorganizing spaces and functions. In the docks area, a new building housing the shopping promenade has been built, with the six cinema halls on its top floor. The whole building complex alternates the respect for the tradition of the city, with the most innovative design. The indications given by the customer to the acoustic consultants were very clear, in the sense of providing an excellent acoustics quality, complemented by the latest technology equipment, including the new Atmos format of the Dolby Laboratories Inc. Starting from these premises, the following tasks were developed: a soundproofing project; a HVAC systems project with perfect accordance of the acoustic requirements, and an acoustic treatment project, developed using the AFMG's EASE software. The project with its construction details, the model executed on software and the main measurements performed are presented.

Architectural Acoustics for Non-Performance Spaces:
[Paper ICA2016-572]

Acoustics of the Border Cultural Centre in the neighbourhood of Palermo, city of Buenos Aires, Argentina

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Abstract
The independent theatre, i.e. the one formed by groups of actors without the representation of an entrepreneur, has a long history in Buenos Aires. In addition to these cultural expressions, numerous musical groups and other disciplines are added to form a diverse and widespread artistic expression. A group of small halls is concentrated within the *Palermo* neighbourhood of Buenos Aires, designed to accommodate these artistic expressions. However, there was still no hall model located in a narrow terrain lot which allows performing simultaneous events, such as acting, music and dance, in the same building without interfering aurally with each other and fulfilling the noise regulations of the City. The projected room came to respond to that need. The architectural design of the cultural centre, which begins from spaces shaped by natural light, with large openings to the outside, was a challenge for the acoustic consultants. The acoustic project carried out for the theatre is presented, where the various solutions adopted to meet the acoustic requirements are shown, respecting and completing the decisive architectural concept.

Friday morning, 9 September 2016 — **Sala de Profesores**
09:00 - 10:40
SOBRAC Meeting

Friday morning, 9 September 2016 — **Sala de Profesores**
11:00 - 12:40
AdAA Meeting

Friday afternoon, 6 September 2016 — **Juan Pablo II Auditorium**
14:30 - 15:50
Architectural Acoustics - Room and Building Acoustics
AA7 - Isotropy and Diffuseness in Room Acoustics

Isotropy and Diffuseness in Room Acoustics:
[Paper ICA2016-232]

Evaluation of temporal diffusion of room impulse responses by using the autocorrelation analysis

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Abstract
This study investigates the autocorrelation function (ACF) of room impulse responses to test the randomness of the temporal distribution of the reflections. The room impulse response ACF can detect strong specular reflections or periodic reflections which cause the perception of tone coloration. The sound fields of two concert halls were compared. These two halls have similar room shape, room volume, and reverberation time, but one has an array of circular column diffusers and another does not. The room acoustic parameters (the initial time delay gap ITDG and the interaural cross-correlation $IACC_{E3}$) already showed difference between these two halls, reflecting the effect of diffusers on the spatial distribution of reflections [Fujii et al. (2004) J. Temporal Des. Architect. Environ. 4, 59-68]. This study further calculates the room impulse response ACF and examines whether it can clarify the

degree of diffusion at each seat. The comparison between the two halls with and without the diffusers showed that the effect of the diffuser was found on the temporal diffusion Δ in 500 Hz and 1000 Hz octave bands, corresponding to the dimension of the diffuser.

Isotropy and Diffuseness in Room Acoustics:
Paper ICA2016-551

Experimental investigation on varied degrees of sound field diffuseness in enclosed spaces

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Abstract
Sound field diffusion in enclosures should be experimentally quantified based on measured room impulse responses. A parameter, the sound field diffusion coefficient (SFDC) is under development. This parameter includes relative global crossover time and its standard deviation of their value over frequency bands. The SFDC expresses the reflection's amplitude control and temporal distribution uniformity, using both broadband and third octave-band energy-decay compensated impulse responses and taking reference with those parameters from a set of impulse responses synthesized with Gaussian white noise. In an attempt to demonstrate the quantification capability of the SFDC, a systematic investigation is conducted whereby varied room configurations using carefully designed scattered interior surfaces are examined with the hypothesis that varied degrees of surface scattering will ultimately lead to varied degrees of sound field diffusion in the enclosure. To this end, a scale-model room is established with interior surface configurations ranging from totally plane surfaces to diffusely reflecting surfaces that cover large portions of the enclosure's interior area. This paper discusses the experimental design and evaluates the results of data collected using systematic modifications of varied degrees of surface scattering, each with combinations of different source orientations and microphone positions.

Isotropy and Diffuseness in Room Acoustics:
Paper ICA2016-574

Gaussian white noise impulse responses as absolute diffusion reference values

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Abstract
A new parameter, the sound field diffusion coefficient (SFDC) is under development. The SFDC expresses the reflection's amplitude control and temporal distribution uniformity in one number, using both broadband and third octave-band energy-decay compensated impulse responses. In order to absolutely quantify the diffusion of a sound field through the SFDC, a new set of reference values is presented. From now on, the implementation of these absolute values will allow the comparison of results between different rooms, independently of their crossover times and room volumes. Based on the fact that information after the crossover time, namely the reverberation tail of every impulse response, can be modeled as an exponentially decaying Gaussian white noise, a group of 24 synthetic, impulse responses were generated under these conditions. Then the mean values of amplitude control and reflection's distribution uniformity, over third octave frequency bands were extracted to configure a new set of diffusion references, which from now on will be associated to the SFDC results for diffusion quantification in terms of Gaussian White Noise units (GWNu).

INVITED

Isotropy and Diffuseness in Room Acoustics:

Paper ICA2016-393

Reproduction of the modal response of enclosures by means of interactive auralization

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Abstract

A method to create an interactive auralization of the modal response of a room is presented. The process is based on the numerical estimation of the spatial impulse responses of the enclosure using a combination of the finite element method and geometrical acoustics. The acoustic field is then reconstructed by means of a plane wave expansion, which allows for interactive features such as translation of the sound field. The auralization is presented to the listener using a headphone-based binaural system. Compared to techniques based only on geometrical acoustic predictions, this hybrid methodology produces a more accurate rendering of the acoustic field at low frequencies, thus providing an effective tool to reproduce the modal response of enclosures in real-time.

Friday afternoon, 6 September 2016 **Dr. Valsecchi Auditorium**

14:30 - 15:10

Soundscape

SS3 - Soundscape, Psychoacoustics and Urban Environment

INVITED

Soundscape, Psychoacoustics and Urban Environment:

Paper ICA2016-267

Influence of experimental conditions on sound pleasantness evaluations

Pierre Aumond[(a)(b)], Florent Masson[(c)], Leandro Beron[(c)], Arnaud Can[(a)], Bert De Coensel[(d)], Dick Botteldooren[(d)], Carlos Ribeiro[(e)], Catherine Lavandier[(b)]

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Abstract

Being able to characterize and estimate the urban sound perception is a key point to improve the city dwellers environmental quality. In the past decade, various studies have focused on collecting perceived global sound pleasantness at specific locations. Some of them were carried out on field in order to evaluate the soundscape perception of the participants directly in their context. Other studies were realized in laboratory to better control the stimuli and to increase the number of participants who were subjected to the same sound environment. Most of the laboratory experiments are done in large or semi-anechoic chamber with calibrated and highly realistic audio reproduction in order to respect the ecological validity of the experiment. On one hand, even with a high immersive level, the laboratory context is not as rich as the field context and the two types of experiment could lead to different results. On the other hand, few studies exist showing the influence of decreasing ecological validity for the same experience. This work presents a short statistical analysis of perceptive evaluations of ten urban locations under 4 different test conditions. First, evaluations are carried out directly in-situ in the city of Paris. Then audio-visual recordings of these locations are evaluated in three different experimental conditions: (i) in a well-controlled acoustic laboratory in Paris region with French people, (ii) in an acoustic laboratory in Buenos Aires with Argentinean participants and lowest

immersive conditions, (iii) in a habitational room with Argentinean participants and subjective calibration. The study reveals that both the “country” factor and the experimental conditions in laboratory do not show any significant impact on the perceived sound pleasantness and perceived loudness assessments.

Soundscape, Psychoacoustics and Urban Environment:
Paper ICA2016-794

Studying urban auditory experiences of Dutch natives in relation to their activities in outdoor public spaces: A proposed methodology

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Abstract
Current studies in soundscape research indicate that the relationship between users of spaces and their soundscapes is influenced by users’ activities as well as their characteristics (including age). However, few studies have focused on researching this influence in a systematic manner. Based on psycholinguistic theories on the existence of two auditory strategies (*holistic hearing* and *descriptive listening*), this paper describes a methodology for getting a deeper understanding of the effect of activity on the description and evaluation of their soundscapes by means of a soundwalk performed with participants of various age groups in a number of public spaces in Amsterdam. The proposed procedure is as follows: we divide participants recruited beforehand into four groups and ask them to perform an activity combining different levels of social interaction and dynamism: walking and talking, sitting and talking, sitting alone and walking alone. We afterwards ask the participants to complete an on-site questionnaire (in Dutch), combining free responses with scales, that allows users to describe and evaluate their soundscapes during their activity, and also provide demographic and personal data. We briefly discuss a data analysis strategy, combining statistical tests with a linguistic analysis of the written corpus (focusing on syntax, morphology and semantics). The methodology described in this paper can contribute to studies that support the process of making cities cater to the needs of more diverse groups of users, by focusing on urban dwellers of various ages and offering insight into their urban auditory experiences.

Friday afternoon, 6 September 2016 — **Cardenal Pironio Auditorium**
14:30 - 15:30
Noise: Sources and Control
NS6 - Noise: Sources and Control (others)

Noise: Sources and Control (others):
Paper ICA2016-116

A field investigation on the vibroacoustic impact of an underground metro line

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Abstract
The paper presents the approach used in a field investigation on the vibration impact of a new underground metro line, focusing on potential annoyance to the population. The metro consists of a double barrel tunnel constructed using a TBM at a depth of 13-25 m. The infrastructure uses a slab track system: the rail is fastened to elastically supported sleeper blocks embedded in a floating concrete slab, cast on a resilient mat. Vibrations were measured over four weeks inside the metro tunnel and inside a selected sample of buildings. Measurements in the metro tunnel were executed applying the sensors on the rail, floating slab and tunnel ring wall. Transits were monitored with video recordings. Measurements in the buildings were done placing the accelerometers on the floor of selected rooms in each investigated building. Approximately 30000 vibration recordings were

collected. A systematic approach was applied to identify the cases in which the vibrations detected inside the buildings could be definitely attributed to metro transits with a high degree of confidence. This task is critical due to the highly variable properties of the soil and of building structures, and to the presence of diverse vibration sources such as surface transportation and building technical systems. The dispersion of the results due to the different characteristics of the rolling stock was also investigated. The results of the study provided a basis for implementing mitigation measures at the sites in which the vibration levels exceeded the limits prescribed by applicable standards.

Noise: Sources and Control (others):

Paper ICA2016-276

Shielding the source

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Abstract

Active noise control is a technique used to attenuate the noise at low frequencies. The basic algorithms need to feedback the pressure at receiver location in order to ensure the attenuation. It implies the silent zone is located very near to the feedback sensors. In order to increase the silent zone at other locations, it has been proposed several methods as virtual sensing and active shielding. The limitations of them are related to the computational cost or the number of sensors and actuators. This article proposes to reduce the number of actuators and sensors using active shielding around the source instead of active shielding around the desired silent zone. It implies to reduce the perimeter of the area reducing the number of sensors and actuators. As a result of this method, it is achieved controlled locations farther than the set of sensors and actuators. This is proven through a discretization of the wave equation. A simulation case shows the good behavior of this proposal.

Noise: Sources and Control (others):

Paper ICA2016-773

Acoustics of tearing Velcro

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Abstract

The distinctive sound of a tearing Velcro has been investigated experimentally. We have constructed a rig that can tear a Velcro in a repeatable way at specified speeds. Tearing leads to the snapping of the hooks and loops of a Velcro, which induces vibrations of the base on which the Velcro is attached. We show that this is the main source of noise and present some new designs to reduce Velcro sound.

Electroacoustics and Audio Engineering:

Paper ICA2016-230

Estimation of directional localization of sound from reproduced wave surface

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Abstract

In this paper, a novel approach to generate a localization curve is proposed. The localization curve, which is a mapping from the angle of a sound source in the recording to that of the virtual image in the reproduction, is widely used in the sound recording. It was usually calculated based on the function that approximates a so-called "phantom source shift" by interchannel level and time differences. As the result, it has been applicable only to the sound from the frontal direction at the ear height. The proposed approach generates the localization curve such that the apparent direction of a sound in the reproduced sound field is estimated by the direction normal to the wave surface and calculates a mapping from the direction of a sound in the recording to the estimated direction. The experimental result showed that, for a two-channel stereo, the new approach generated a localization curve that was similar to the curve by the conventional method. Moreover, the new approach generated the curves for 5.1 channel and 22.2 channel systems. Since the arrangement of channels in the 22.2 channel system biases to the top layer (there are 9 channels in the top layer and 3 channels in the bottom layer), the vertical localization curve obtained by the proposed approach biased to the upper direction, suggesting that the sound image tends to rise in the natural recording of 22.2 channel.

Electroacoustics and Audio Engineering:

Paper ICA2016-422

Photoliptophone: A virtually unknown ancestor of optical audio systems to reproduce printed sound on plain paper

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Abstract

The Photoliptophone was a system developed by Argentinean inventor Fernando Crudo in the 1930s. The system allowed recording audio waveforms on radiographic plates, so that they could be printed on paper. The aim of the system was reducing the distribution cost of audio copies by including the printed waveforms in magazines and newspapers. The Photoliptophone obtained patents in thirty countries. National and international reports evaluating the operation of the prototype describe the Photoliptophone as a system with good audio quality. The system remained in operation for two decades, although it never managed to successfully enter the market as an alternative to mechanical recording systems. In the same period of time, a recording system known as Selenophone was developed in Austria. It used 7mm film to record audio and which could be printed on reels of paper to be later reproduced as sound. The aim of the system was to record over longer time periods than mechanical systems allowed for, as well as printing copies on paper to reduce the cost of film as physical support. This paper describes the operational principles of the Photoliptophone and compares it to the Selenophone. Original documents related to the Photoliptophone have been analyzed for this study. Such documents include patents, inventor's personal letters, official use permits, business plans and audio waveforms printed on either paper or radiographic film. An evaluation of the audio quality of the Photoliptophone is also reported in this paper. Original printed audio waveforms that remained untouched for the last eighty years have been scanned. The associated audio was retrieved by means of numerical processing of the scanned images.

Electroacoustics and Audio Engineering:

Paper ICA2016-559

Plate reverberation: Towards the development of a real-time physical model for the working musician

Michele Ducceschi[a], **Craig J. Webb**[b]

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Abstract

Reverberation is an essential effect for sound design and music production, and commercially available software offers an unprecedented range of solutions for artists. Plate reverberation represents an attractive choice as the modal density of large plates is constant over the audible range, creating a uniform response in the frequency domain. However, available plug-ins rely on either sampled impulse responses or simple delay algorithms. In this paper, a pure physical model of a rectangular plate is used at the core of a real-time effect plug-in. The user has a choice of intuitive parameters to select the dimensions of the plate, the tension, and decay ratios. The input and output positions can be changed dynamically, during the runtime of the simulation. This represents a clear improvement over static algorithms based on impulse responses. Optimisation of the model using cpu vector intrinsics is demonstrated, allowing real-time computation of large scale plate reverbs on consumer hardware using a standard plugin architecture.

Electroacoustics and Audio Engineering:

Paper ICA2016-613

Effects of operation at and off-electrical resonance on the performance indices of linear alternators under thermoacoustic-power-conversion conditions

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Abstract

Thermoacoustic power converters consist of thermoacoustic engines that convert thermal energy into acoustic energy and linear alternators that convert the generated acoustic energy into electric energy. The conditions required for best acoustic-to-electric power conversion include that linear alternators operate under mechanical and electrical resonance simultaneously causing the acoustic impedance of the linear alternator to become purely real. Electrical resonance is achieved by balancing the linear alternator inductor's impedance by using a power-factor-correcting capacitor. However, the exact capacitance value depends on the mechanical stroke, which in turn depends on the load seen by the linear alternator, including the value of the capacitance used. Thus, if operation takes place at off-design conditions, the mechanical stroke in operation and the capacitance used may not lead to electrical resonance. This work experimentally investigates the linear alternator performance indices, namely the mechanical stroke, the dynamic pressure at the face of the linear alternator's piston, the output electric power, the generated volt, the generated current, the acoustic-to-electric conversion efficiency, the mechanical-motion loss, the Ohmic loss, and the fluid-seal loss when operating at electrical resonance and when operating at different levels of off-electrical resonance for two types of loads: a linear (resistive load) and a non-linear constant-voltage DC electronic load. Increases in the acoustic to-electric conversion efficiencies of up to 27.8% and 54.7% can take place when operating at electrical resonance in the linear and non-linear cases, respectively. The effects of operation at and off-electrical resonance conditions on the harmonic generation and on the acoustic impedance under linear and non-linear loadings are presented.

String Instruments:
Paper ICA2016-8

About the acoustic and other non-destructive methods for the characterization of old historical string musical instruments – an overview

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Abstract
Old historical string musical instruments in Western cultural tradition have been recognised as objects of universal cultural heritage since 1967. These instruments are those from the violin family, guitars, harps, harpsichords and pianos built since before the 14th century. These rare and unique musical instruments specifically addressed vital questions related their use, restoration and conservation. For the characterization of historical musical instruments one or several non-destructive tests could be selected depending on the purposes of the investigations, namely the characterization of their materials or the behaviour of the entire instrument.

INVITED
String Instruments:
Paper ICA2016-25

Comparison between three different Viennese pianos of the nineteenth century

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Abstract
Measurements are performed on three pianos built in Vienna during the 19th century by three generations of the Streicher family. These selected pianos are representative examples of the evolution of Viennese piano making. The first piano (NS19) was made in 1819 by Nanette Streicher: its structure is close to an harpsichord, with a thin soundboard and a single bridge. The second piano (JBS36) was built by her son, Johann Baptist, in 1836. Its soundboard has wider ribs, and its bridge is divided in two parts. The string scaling shows higher tensions compared to NS19. Finally, the third piano (JBSS73) was made by Emil Streicher, Johann Baptist's son, in 1973. This piano is larger than the two others. Its soundboard is thicker, again with an increase of tension compared to JBS36. Physical parameters relative to the geometry and material of the constitutive parts of the pianos (strings, hammers, soundboard) are derived from these measurements. These parameters serve here as input data for simulating vibrations of strings and soundboard, thanks to a time-domain model of a piano (Chabassier et al., Jasa 134(1), pp. 648-665) which couples together the hammer, the strings, the soundboard and the acoustic field. Fine adjustments of the parameters are made, by comparing measured and simulated waveforms. Further simulations are conducted with systematic variations of selected parameters (hammer mass, string tension, soundboard thickness,...). These variations, which would be hard (or even impossible) to achieve in the reality shed a new light on the links between physical parameters and sound quality, with an historical perspective on the art of piano making.

String Instruments:
[Paper ICA2016-171]

Contribution of the vibration of various piano components in the resulting piano sound

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Abstract
To date, piano sound modelling is focused primarily on the vibrational behaviour of the strings and soundboard. However, it is observed that other components of piano such as the rim also vibrate when the piano is being played. Current work serves as a pilot experimental investigation on the contribution of the vibration of various components of the piano to the resulting piano sound. The components inspected are the soundboard, the inner and outer rim, the cast-iron frame and the lid. Vibrations of the components are captured by accelerometers and in parallel, sound pressure is recorded by microphones. Operational transfer path analysis is conducted to identify the main contributors of the sound.

Friday afternoon, 6 September 2016 **Auditorium 2**
14:30 - 15:50
Psychological and Physiological Acoustics
PP1 - Free-Field Virtual Psychoacoustics and Hearing Impairment

Free-Field Virtual Psychoacoustics and Hearing Impairment:
[Paper ICA2016-493]

Impact of spatial audiovisual coherence on source unmasking

Julian Palacino[(a)], **Mathieu Paquier**[(a)], **Vincent Koehl**[(a)], **Frédéric Changenet**[(b)], **Etienne Corteel**[(c)]
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Abstract
The influence of the spatial audiovisual coherence is evaluated in the context of a video recording of live music. In this context, audio engineers currently balance the audio spectrum to unmask each music instrument getting it intelligible inside the stereo mix. In contrast, sound engineers using spatial audio technologies have reported that sound source equalization is unnecessary in live music mixing when the sound sources are played at the same location of the physical instruments. The effects of spatial audiovisual coherence and sound spatialization have been assessed: expert subjects were asked to compare two mixes in audio only and in audiovisual mode. For this aim, music concerts were visually projected and audio rendered using WFS. Three sound engineers did the audio mixing for all pieces of music in the same room were the test have been carried out.

Free-Field Virtual Psychoacoustics and Hearing Impairment:
[Paper ICA2016-775]

Does learning a room's reflections aid spatial hearing?

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Abstract
Sound reflections are abundantly present in everyday environments; yet, our spatial hearing abilities are usually not impaired by them. One contributor to this robustness is the adaptation to the reflections after being repeatedly exposed to the room's reverberation. The echo threshold, the delay at which a reflection starts being separately audible as an echo, increases with repeated exposure to the reflection

pattern. Benefits from prior exposure to a room's acoustics have also been shown for speech understanding. Here we study if learning the characteristics of the room's reverberation pattern can improve sound localization. Stimuli were presented in the free-field of the Simulated Open Field Environment (v3), a room acoustics simulation and auralization tool based on the extended mirror-image source method with auralization over 96 loudspeakers. Participants localized target noise bursts presented in the front of the listener in a virtual room either with no prior information about the room, after a short exposure phase consisting of two noise bursts presented in that room, or after a long exposure phase consisting of 14 noise bursts. The exposure stimuli were presented from random locations for each burst at the sides in order to transmit information about the room, but to prevent interference with the target stimulus locations. Localization ability as measured by RMS error and standard deviation was improved after prior exposure to the room acoustics. Results indicate that learning room acoustics can aid our ability to locate sound sources in rooms. Since exposure and target stimuli did not share the same positions and reflection patterns, results also demonstrate a generalization within the room in that the improvement from room learning can carry over from one to another position.

Free-field Virtual Psychoacoustic and Hearing Impairment:

Paper ICA2016-373

Prior exposure to room acoustics and its effect on localization

Samuel W. Clapp[(a)], Bernhard U. Seeber[(a)]
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Abstract

In the majority of studies of psychoacoustics in rooms, test subjects do not interact with the stimuli presented to them. One form of interaction that has been increasingly deployed in recent studies is head movements, which have been shown to improve realism and reduce front-back confusions. Recent research and advances in computing power have paved the way for new forms of interaction, making it possible for subjects to modify their own position and that of sound sources in a virtual room, with fast updates to the room simulation. In this study, two conditions are tested in order to examine how localization judgments of a short speech target in the presence of a noise distracter are affected by prior experience of a room's acoustics. In the first condition, no prior exposure to the simulated room's acoustics is given. In the second condition, listeners are able to first explore the room by manipulating the source position within the room geometry and hearing the resulting auralizations, before being tested again with the same target stimuli. These conditions test the hypothesis that active exploration and learning of a room's acoustics can be accomplished with practice, and can assist in source localization in non-ideal conditions. Results show large localization errors occur without prior exposure to the room's acoustics, always pulled in the direction of the noise source. Reduced localization errors are seen following the interactive exploration session. In both conditions, errors are larger the farther away in azimuth the target is located from the noise source.

Psychological and Physiological Acoustics (others):

Paper ICA2016-400

Analysis of human response to combined noise and vibration in airplanes

Júlio Alexandre de Matheucci e Silva Teixeira[(a)], Roberto Jordan[(b)]
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Abstract

A mock-up was designed to simulate an internal airplane environment, with the objective of performing subjective analyses of inflight noise and vibrations recorded signals. This simulator physically reproduces an aircraft section, with only one centralized seat, projected in this way to reduce as much as possible the coupling among seat movements. Sounds are reproduced by a high fidelity earphone and the vibrations are provided by a triaxial set of shakers, positioned under the seat. Sound and vibration signals were recorded in cruise flights, in various seat positions. Binaural sound signals were obtained with the help of an artificial torso and a triaxial accelerometer, positioned near the seat attachment point, at the floor, was responsible by the vibration signals generation. Some signals had their vibrations and their noise amplified and the other reduced by 3 dB, creating a set of 30 stimuli. As

the resulting set of signals was too large to be analysed in only one subjective test, six groups (subsets) of combined noise and vibrations signals were created. Two subjective types of analysis were applied: semantic differential (SD) and response scale (RS). SD evaluations are based on opposite adjectives (like “comfortable/uncomfortable”) and when using RS different grades of an equivalent substantive (like “comfort”) are applied to the signals. Some initial analyses were performed to detect if the age, gender or number of recent flights of the subjects influence somewhat the responses at the tests. Results of combined applications of sound and vibrations to subjects are fully analysed and commented, with the use of statistical tools. At the end general conclusions about all performed tests are presented.

Friday afternoon, 9 September 2016 **Auditorium 3**
14:30 - 16:10
AJA Group Meeting

Friday afternoon, 6 September 2016 **Juan Pablo II Auditorium**
15:50 - 16:50
Plenary lecture
Chair: Antonio Pérez-López

Samir Gerges

Paper ICA2016-895

Hearing Protectors: State of the Art and Emerging Technologies of Comfort and Uncertainty in Measurements

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Abstract
In many industrial and military situations it is not practical or economical to reduce ambient noise to levels that present neither a hazard to hearing nor annoyance. In these situations, personal hearing protection devices are capable of reducing the noise by up to around 35 dB. Although the use of a hearing protector is recommended as a temporary solution until action is taken to control the noise, in practice, it ends up as a permanent solution in most cases. Therefore, hearing protectors must be both efficient in terms of noise attenuation and comfortable to wear. Comfort in this case is related to the agreement of the user to wear the hearing protector consistently and correctly at all times. The purpose of this paper is to review the stat of art for the need to develop methods to quantify comfort and noise leakage, also to quantify the uncertainty in evaluating hearing protector noise attenuation.

Friday afternoon, 9 September 2016 **Foyer Juan Pablo II**
Closing Ceremony
16:50 - 17:30

Friday afternoon, 9 September 2016 **Foyer Juan Pablo II**
Farewell cocktail
17:30 - 18:30

FIA 2016 / AdAA 2016 / SOBRAC 2016 ABSTRACTS

Tuesday, 6 September 2016

Tuesday morning, 6 September 2016 **Microcinema**
09:00 - 10:20
FIA-NS - Noise: Sources and Control
FIA-NS - Ruido: Fuentes y su Control
FIA-NS - Ruído: Fontes e o seu Controlo

Noise: Sources and Control:

Paper FIA2016-6

Analytical and numerical estimation of acoustic transfer function in ducts considering general boundary conditions

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Abstract

Many engineering systems in operation produce undesirable noise which may attenuate by the use of proper noise control technique. Classical examples, present in the human life, are: automotive exhaust; heating, ventilating, and air conditioning (HVAC) equipment. The main component of these examples is the duct. Nowadays, many noise control techniques are available for acoustic ducts: passive and active methods. For both techniques, it is necessary to know very well the acoustic behavior of the system in order to have good and effective results. Acoustic duct has intrinsic characteristics which make it complex, like no natural roll-off at high frequencies and it is modally full. So an accurate acoustic model for use in design stage is very important to obtain success in the noise control implementation. Most researches use experiments to identify the acoustic duct model. In order to reduce cost and time, analytical and numeric models is a good alternative. In this work the acoustic transfer functions (TF) between noise source and microphone located inside the duct are estimated for some configurations and general boundary conditions. Analytical TF for linear, time-invariant and infinite-dimensional acoustic duct is presented in frequency domain derived from fundamental wave equation and using Laplace transform. In addition, finite element model is developed for comparison. Both models showed good approach, although the open boundary condition adds some differences between the TFs when frequency increases.

Noise: Sources and Control:

[Paper FIA2016-100]

Experimental study of the relationship between radiated sound and machining conditions of a wood shaper machine based on acoustic camera measurements

José Luis Barros[a], Alfredo Aguilera[b]

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Abstract

The current work discusses the relationship between sound radiation and machining conditions (i.e. head rotation frequency, feed speed, etc.) for a wood shaper machine. An acoustic camera was used to generate acoustic maps of radiated sound for different cutting conditions. It was determined that there are four main sources of noise emission in the shaper machine, which consist of the suction airflow system, the automatic feed system, the cutter headdrive system, and the sound associated to the cutting tool in contact with the wood to be processed. The acquired acoustic signals were filtered in order to eliminate signals caused by external sources not directly related to the cutting process and frequency components that do not contribute to the differentiation of machining conditions. The resulting acoustic maps for the different cutting conditions are shown and the feasibility of using this acoustic maps as a tool of machining conditions monitoring is analysed.

Noise: Sources and Control:

[Paper FIA2016-116]

Acoustic properties of recycled textile materials

Juan Manuel Loria[a], Leonardo Magliolo[b], Joaquín Mansilla[c], Alejandro Bidondo[d] Nicolás Urquiza[e], Gonzalo Botto[f], Gabriel Santiago Rosanigo[g]

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Abstract

Experimenting with industrial discards, mainly textiles materials, allows to develop decorative pieces with sound absorption properties. In order to improve comfort, these elements can improve acoustic conditioning in various spaces of different use (offices, homes, schools, bars, etc.). With discards recovered from the industry as prime material, the company Feboasoma has designed different acoustic absorbent materials as decorative products. The international standard ISO 354 is used to calculate the sound absorption coefficients (α) of different products, as other materials as yarn and fabrics polar type, and experimentation on new materials of natural products as wood chips of pine, privet and several tree roots. The measurements were made in two different halls adapted as reverberant chambers, one located in the company Feboasoma and another in the "Universidad Nacional Tres de Febrero" (UNTREF), both in Buenos Aires, Argentina. The results allowed to evaluate the efficiency of recycled materials to be used for acoustic purposes without neglecting aesthetics.

Noise: Sources and Control:

[Paper FIA2016-102]

Sound absorption coefficient measurement based on the Transfer Function Method specified by the standard ISO 10534-2 using low-cost alternatives

Thaynan Oliveira[(a)], Paulo H. Mareze[(a)], Matheus Pereira[(a)], Sergio Aguirre[(b)], William D'A. Fonseca[(a)], Eric Brandão[(a)], Rogério Pirk[(c)]

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Abstract

Porous materials are usually employed since they are a well-known and effective passive approach. This class of material converts the acoustical energy into heat by viscous and thermal effects. That is, this transduction is carried out by the particles' friction and movement inside the pores. Considering this scenario, it is clear that characterization of porous materials plays an important role in noise control. A common way to estimate their properties is via Kundt's Tube measurements, where the sound absorption coefficients are extracted by applying a normal incidence acoustic wave over the sample. The method is standardized by the ISO 10534-2, which recommends Class 1 instrumentation. This work presents two groups of measurements. The former case is the fruit of a low-cost measurement system, applied to a custom Kundt's Tube constructed in the Acoustical Laboratory of the Federal University of Santa Maria, in south Brazil. In the latter case, a commercial Kundt's Tube together with Class1 instrumentation is applied to the approximately same situation. The results are discussed and the analyses point out that the developed low-cost system has a satisfactory performance (considering the frequency range studied), achieving similar results.

Tuesday morning, 6 September 2016 **Microcinema**
10:20 - 10:40
FIA-PP - Psychological and Physiological Acoustics
FIA-PP - Acústica Psicológica y Fisiológica
FIA-PP - Acústica Psicológica e Fisiológica

Psychological and Physiological Acoustics:

[Paper FIA2016-37]

Noise on an aircraft cabin: Effect of power density spectrum on the noisiness

Hugo Scagnetti[(a)], Shin-ichi Sato[(b)], Florent Masson[(c)]

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Abstract

The aim of this paper is to determine the correlation between different acoustic descriptors related to power density spectrum and perceived noisiness in sounds recorded inside a propeller aircraft cabin. Particularly, it is sought to demonstrate a strong correlation between tonal components of a signal and its noisiness. For this purpose, a paired comparison test was conducted with 26 test subjects and 8 stimuli. In order to exclude the effect of the sound pressure level on the noisiness, the L_{Aeq} of all the signals used in the test were adjusted to 64 dBA. The results of the analysis showed a strong correlation between the numbers of tonal components with the perceived noisiness. It was also found that the amplitude of the 500 Hz tone has a great contribution to the perception of noisiness. Moreover, it was found that at higher frequencies that 3000 Hz, the amplitude of the tonal components has a lesser effect on the perception of noisiness.

Psychological and Physiological Acoustics:
Paper FIA2016-86

Measurements, modeling and analysis of industrial noise in the city of coronel (Chile) – Advantages of using acoustic camera for source location

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Abstract
The current work discusses the assessment of industrial noise sources in the city of Coronel, Chile. Such evaluation is part of the study 'Assessment of Noise Levels in Coronel' developed by the Chilean Ministry of Environment. The research presented here discussed the methodology used for the acoustic modeling of industrial noise and advantages of measuring with an acoustic-camera (beamforming technique) in order to locate primary sound sources. Measurements of noise primary sources in the south of the city were carried out using acoustic-camera in order to establish a set of point sources with its corresponding sound power spectrum. Using the generated data, the noise emissions of different industrial plants existing in the area are modeling. The resulting industrial noise maps are shown and the impact of each specific industry on the community is analysed from the point of view of noise pollution.

Psychological and Physiological Acoustics:
Paper FIA2016-26

Annoyance of industrial noise with tonal component at different frequencies

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Abstract
Several previous experiments about tonal noise found that the annoyance varies when the frequency of the tone changes. In the present work a subjective test on the annoyance of industrial noises with tonal components were performed. A non-tonal sound from a textile factory has been used for the test and modified into 8 tonal noises in each octave band from 63 to 8000 Hz. In this work the definition of tonal noise from the standard ISO 1996-2 has been used. This study conducts a paired comparison test and analysed the variation of annoyance for these tonal noises. Each noise was set to an equivalent sound pressure level (SPL) of 85 dBA for the subjective test. The results of the research showed that, regardless of frequency, the presence of the tonal component makes the sound more annoying. Furthermore the greatest annoyance occurs at 4 kHz, where the ears are the most sensitive.

Psychological and Physiological Acoustics:
Paper FIA2016-28

Concentration mapping of noise pollution complaints in Natal/RN (Brazil) between 2012 and 2015

Luciana Alves[a], Tamáris Brasileiro[b], Renata Araujo[c], Débora Florêncio[d], Virgínia Araújo[e], Bianca Araújo[f]
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Abstract
People's individual actions can impact a whole community. It is possible to infer that the population, habitually, disregards that the environmental noise directly affects itself and often produces it above the desirable noise levels considered for acoustic comfort and health. The noise pollution above these levels is classified as noise and constitutes an environmental impact, very evident in big cities, being computed by government agencies through complaints made by the population. Thus, this study aims to analyze the situation of the noise pollution in Natal/RN/Brazil, in order to understand the origin of those events and seek to identify tools to support municipal management in a most effective combat to noise pollution. This article is the completion of a study conducted to the years 2012 and 2013, in order to identify the evolution of the types, location and number of complaints. Reports of noise pollution made by the population -during the years 2014 and 2015- were collected at Secretaria Municipal de Meio Ambiente e Urbanismo (SEMURB). These complaints were classified according to the generating factor and the sound source, spatialized punctually at Natal's map and analyzed by the concentration of sound levels in the acoustic mapping software SoundPLAN®. Through the data collected and generated maps, it is concluded that Natal exceeds regulatory limits on urban noise throughout the city, whereas the highest rate of complaints is from existing bars, however the rates worsen in commercial districts, regardless of the generating factor. In the studied space-time, there is a similar behavior, but growing in complaints. This indicates that it is necessary, in addition to coercion actions, environmental education for the population with regard to noise pollution.

Tuesday afternoon, 6 September 2016 **Microcinema**
14:30 - 16:10
FIA-EN - Environmental Acoustics & Community Noise
FIA-EN - Acústica Ambiental y Ruido Comunitario
FIA-EN - Acústica Ambiental e Ruído Comunitário

Environmental Acoustics & Community Noise:
Paper FIA2016-29

Concentration mapping of noise pollution complaints in João Pessoa-PB (Brazil) between 2012 and 2015

Tamáris Brasileiro[a], Luciana Alves[b], Renata Araujo[c], Débora Florêncio[d], Virgínia Araújo[e], Bianca Araújo[f]
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Abstract
The high level of urban noise has increased the index of noise pollution in big cities. Point sound sources contributes to the increase of this kind of pollution, compromising the quality of life of the population, active and passive agent of this problem. The population has the complaint to the inspection municipal agencies as the best means of procedure against noise pollution. This research aims to conduct a study

of spacialization and impact of the concentration of noise pollution complaints in the Capital city of João Pessoa/PB (Brazil) in the past four years (2012 until 2015). For this, it has been collected in the Secretaria do Meio Ambiente (SEMAM) and the Superintendência de Administração do Meio Urbano (SUDEMA), data noise complaints made by the population during the above period. After separating them by type of use of the edification and sound source, the complaints were located on a map of the city, through the geographic coordinate system, and inserted with the estimated sound levels in the acoustic mapping software SoundPLAN®. In this way, through a simplified mapping, it is possible to verify the impact of the concentration of high sound levels indicators, considering only the sound distribution for each point sound source. Through this mapping, it was possible to identify that noise pollution in João Pessoa/PB is entirely evident considering only the reported point sources. The city largely exceeds the limits established by the regulations, being the majoritycomplaints established through the annoyance between neighbors. In addition to this, when it comes to location and type of source emitted, the city has the same concentration profile of complaints among the four years studied.

Environmental Acoustics & Community Noise:

Paper FIA2016-49

The transport stream interfering in noise pollution in comercial and residential districts in the city of Maceó-AL, Brazil

Stella Oliveira[(a)], Ana Caroline Araújo[(b)], Adila Melo[(c)], Adna Oliveira[(d)], Willian Oliveira[(e)], Maria Lucia Gondim da Rosa Oiticica[(f)]

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Abstract

With the growth of cities, residential areas were losing their essence of comfort, being victims of the main causes of the incidence of noise pollution. The heavy traffic and the improper installation of trade points in urban spaces led to the search of transport and marketing spaces for use of their everyday activities. The aim of this study is to assess the participation of the transport flow in noise pollution in neighbourhoods considered commercial and/or residential in the city of Maceió-AL, Brazil. The methodology used consisted of measurements of the sound pressure level, land-use surveys and the flow of vehicles in selected region in various pre-established points. The projects selected were five boroughs (commercial and/or residential) of the lower and central city. This mapping will be observed the impact of road traffic noise in urban areas, giving voice to neighbourhoods and communities, aiming to the right and a better comfort of users in accordance with the existing reality. In the surveys carried out have been found high levels of noise pollution in vast majority close to 78dB (A) in every neighbourhood. Small excerpts of these neighbourhoods have shown quieter areas between 55dB (A) (district of Prado) and 60dB (Pontal da Barra Neighbourhood). The results present will complement the existing seaside database from the city of Maceió, situated in northeaster of Brazil in order to elaboration the acoustic map of the city that can serve as acoustic awareness of the importance of preservation of an environment acoustically healthy.

Environmental Acoustics & Community Noise:

Paper FIA2016-12

Analysis of the influence of public transport in noise levels in the city of Talca, Chile

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Abstract

Noise pollution is a major environmental problem in cities around the world and road traffic is considered the main noise source. The public transport may constitute a major part of road traffic and in the city of Talca (Chile), 60% of commutes are done in public transport. Because of this, the sound contributions from different types of public transport were analysed in this study. To this end, a random set of sampling points were placed in streets with different role in urban connectivity according to the Ministry of Transport and Telecommunications of Chile. Results show significant relations between the registered noise levels and the flow of public transport. Significant differences were also found in the sound signature and in noise contribution in the service road type among the various types of public transport. Public transport produces increases in noise by road type from 0.37 dB in highway road type to 2.05 dB in service road type. Therefore, these results should be taken into account in any action to improve the noise situation or urban connectivity in cities like Talca.

Environmental Acoustics & Community Noise:
Paper FIA2016-88

Characterization of noise pollution in downtown of Cordoba city

Jorge Perez Villalobo[a], Horacio Contrera[a], Raúl Bodoira[a], Elías Cáceres[a], María Hinalaf[a], Mario Serra[a]

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Abstract

Noise pollution in urban areas is mainly generated by the vehicular traffic. In this paper the results obtained from a survey conducted about the levels of noise pollution daytime generated by road traffic in an area with commercial-residential predominance on Cordoba city, Argentina, is reported. The study area has an approximate surface of 1.3 km^2. The measurements were carried out in: (1) major avenues that cross the city; (2) secondary streets with a medium density of vehicular traffic; (3) a street with high flow of passenger vehicles, mainly buses. From the sound levels surveyed in fixed points of the area under study, two groups of noise maps were developed in order to show the extent of noise pollution through of: (a) the overall levels with A weighting and (b) the spectral composition analysed by standard octave bands without weighting.

Environmental Acoustics & Community Noise:
Paper FIA2016-103

Monte Carlo simulations to estimate the impact of vehicle traffic noise dynamics in the vicinity of a bus stop based on real sound signals

Italo C. M. Guedes[a], Stelamaris Rolla Bertoli[b], Jugurta Montalvão[c]

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Abstract

The noise of vehicular traffic is the main source of noise pollution in cities. In urban areas, power fluctuation in this kind of noise is the result of the instabilities in traffic due to crossings, speed bumps, bus stops, and behavior of drivers. The objective of this study is to estimate the impact of vehicular traffic noise dynamics in the vicinity of a bus stop. Simulations were done using probabilistic model based on the Monte Carlo method, considering the real sound signals and randomness of the flow and composition of vehicles and process of arrivals and departures of buses at the analyzed bus stop. The hypothesis of the research is: the variability of noise due to arrival and departure of buses at bus stop increases the level of local noise impact. To this end, we adopted the Traffic Noise Index and the Noise Pollution Level as the main acoustic parameters and the following procedures: selection of the study object – bus stop (Campinas – Brazil); recording real vehicles sound signals; acquisition of geometric parameters, acoustic and traffic cues; representative simulations of traffic noise in different scenarios, by altering the time between successive bus arrivals at the bus stop; statistical analysis of the results. We concluded that the simulation model was sensitive to changes in the time between

successive bus arrivals at the bus stop. Smaller values of the average interval between arrivals were associated with higher levels of noise impact. The simulation method and analysis proposed appear to be a promising tool to evaluate the influence of bus stops in traffic noise, with future possibilities of incorporating the "listening" as another way to subjectively evaluate simulated noise.

Tuesday afternoon, 6 September 2016 **Microcinema**
16:30 - 18:50
FIA-EN - Environmental Acoustics & Community Noise
FIA-EN - Acústica Ambiental y Ruido Comunitario
FIA-EN - Acústica Ambiental e Ruído Comunitário

Environmental Acoustics & Community Noise:
Paper FIA2016-43

Acoustic mapping in the Institute of Biological Science from UFMG: Identification of the noise fonts and sound pressure levels in outdoors areas

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Abstract
This work evaluated the sound pressure levels in external areas inside the Institute of Biological Sciences – ICB on the Federal University of Minas Gerais – UFMG, Pampulha Campus, where students, teachers and employees pass by. The methodology adopted was accorded to the Brazilian standards revision project ABNT NBR 10151 and used Class 1 instrumentation, as accorded with IEC 61672-1, always registering the sound pressure levels in the domain of time and frequency, in bands of 1/3 octaves. The main sources of continuum noise found are related to refrigerators and air conditioning equipment, all located in the laboratories, auditoriums, halls and classrooms. As for the occasional sounds mainly consisted on traffic of heavy vehicles, overflight of aircraft and animal noises. The results were higher than what is recommended for humans. Besides the people who are part of this academic community, ICB has various laboratories with different fauna, also exposed to these noises. This research was motivated by people complaints and uncomfortable impression experienced by the community, which were registered. Also this work could complement further research inside the classrooms, laboratories and auditoriums, all distributed between the four floors from seventeen blocks which constitute the institute. The final results show that there is a demand for solutions in order to achieve the control and suppression of these noise fonts.

Environmental Acoustics & Community Noise:
Paper FIA2016-73

Time representative window for the measurement of urban noise in La Plata city

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Abstract
This work attempts to know a temporal window, throughout a typical day in the city, which is representative of daily urban noise parameters (such as the L_{day}), within a known and acceptable deviation. To do this, a continuous on field sampling of the urban noise throughout 12 daytime hours and without interruption are

carried out. Then is stored the data in a digital form. This procedure is repeated in several locations in the city of La Plata, picking them so that, between them, have different characteristics of urban activity. That can cover, in some way, the different conditions of noise that can be found in the city. Later a laboratory processing is performed, analyzing the behavior of different acoustic parameters for various intervals of measurement and with different durations of time, and performed at different times of the day. Also, the results are compared with those obtained in previous years at the same locations.

Environmental Acoustics & Community Noise:

Paper FIA2016-84

Road traffic noise impact on facades of buildings

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Abstract

Considering that current acoustic scenario of great urban centres has been characterized by excessive noise, mainly noise from road traffic, the function of buildings' facade as sound protection elements has become relevant and must receive attention from designers and builders. In 2013, got into effect the Brazilian Standard (NBR) 15575 – Residential buildings – Performance, the first Brazilian standard that defines requirements to residential buildings' quality considering several aspects, among them acoustic comfort. On its part 4, the standard sets performance requirements for internal and external partitions. The facade composition is particularly important because it is the element that manages the interference between the internal and external environment of the building. For example, the permeability to the wind, to get natural ventilation, which is extremely important as strategy of comfort in tropical regions, depending on the sound environment, can result in overexposure noise in the interior of the buildings. Because facades are elements that have two antagonistic and so important functions – sealing and permeability, they require a careful study during the project. Thus, this paper aims to present a method to analyse the impact of noise emitted by road traffic routes on building facades to be used as a design tool, examining the possibilities of use of natural ventilation without prejudice to the acoustic comfort.

Environmental Acoustics & Community Noise:

Paper FIA2016-98

Guidelines for noise classification methods of residential buildings, according to Brazilian standard ABNT NBR 15575-4

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Abstract

Since the publication of the Brazilian standard ABNT NBR 15575:2013, commonly known as "Performance Standard", there is a growing discussion among specialists about its requirements. Part 4, which deals with internal and external wall systems, indicates required facade performances according to different noise classes, depending on the building location. There are three noise classes, where the document presents non-objective concepts, which give rise to interpretations that may lead to conflicts according to different interests. In 2013, the national association ProAcustica, published a manual of that Performance Standard in order to provide guidelines for its application. In one of those contributions, this manual shows numerical indications of incident noise levels on the dormitories facades, to present more objective definitions of the Noise Classes. Even so, predictive studies aiming to classify future buildings still in the design stage are variable, which may result in differences of application and interpretation of collected data. Thus, there is a lack of recommendations that align the classification methods to the requirements. This paper intends to propose suggestions for good practice on noise classification methods of residential buildings, according to the Brazilian standard, taking into account the building design, its location and other details. It gives emphasis to the importance of considering sound propagation simulation.

Environmental Acoustics & Community Noise:

Paper FIA2016-11

Nocturnal soundscape of Brasilia's Pilot Plan: study case in North Superblock 410 (SQN 410)]

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Abstract

Heritage of Humanity, the Brasilia's Pilot Plan has a distinctive sound ambience, atypical to a big city. When the urban plan was designed by Lucio Costa, noise was not an evident principle, but the adopted solutions incorporated premises that contribute with the urban acoustic comfort - like the existence of local commercial buildings and green belt, which protects the residential buildings. Scientific research carried out for UNESCO identified low percentage of people by traffic noise. However, the nocturnal noise is actually a relevant nuisance factor to the population, causing conflicts between community, bar owners and cultural producers. These annoyances emerged mainly due to the growth of nocturnal activity in Local Commercial Sectors in recent years, and due to their proximity with residential buildings of the Superblocks. By evaluating the soundscape of North Superblock 410 (SQN 410), a place of intense nocturnal activity, we sought to identify and analyze the different sounds that compose the soundscape of that place. The study was conducted based on sound map by computer simulation and in situ measurements, combined with residents' interviews, having a night time frame. The results were analyzed considering both the morphology of SQN 410 and subjective aspects highlighted by residents in interviews. Therefore, we intended to contribute for a better understanding about the acoustic aspects of Lucio Costa's Plan, considering the present uses and the morphological characteristics of the city. From this understanding, possible ways to mitigate conflicts between leisure activities and residential use in this location are pointed out, as well as guidelines for urban legislation.

Environmental Acoustics & Community Noise:

Paper FIA2016-47

Acoustical treatment solution for a large water cooling and chiller system at the rooftop of a commercial building

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Abstract

This article is a case study concerning the operation of 4 chillers, 2 cooling towers and 9 pumps at the rooftop of a commercial building located next to several residential buildings in São Paulo, Brazil. The study involvedsound measurements next to the noise sources (rooftop) and at the vicinity (ground floor), in order to characterize thenoise power of each source and the neighborhood background noise. Data such as topography, traffic at the nearest roads andblueprints of the installation and neighborhood were gathered to develop the digital ground model (DGM) for the sound propagationsimulation´s software. The sound measurements were, then, used to calibrate the digital model - to adjust it so it reproduces thesame noise levels acquired at the site. Once the model was 'calibrated',calculations were performed considering the sound propagation from the noise sources in order to estimate the impact on the nearby residentialbuildings. Then, mitigation measures at the water cooling and chiller systems were simulated in order to meetthe noise criteria defined in the city of São Paulolegislation. The mitigation measures were implemented at the siteand their results confirmed the accuracy of the study developed with the noise simulation software.

Environmental Acoustics & Community Noise:

Paper FIA2016-75

Variation of sound levels in the central area of the city of Buenos Aires after the implementation of priority pedestrian plan

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Abstract

Pedestrian Priority Plan was conceived as a renovation of the downtown's public space project which would cause one of the largest urban variations in recent years. It also gazed out the restructuring and pedestrian of the vehicle routes that spanned more than a hundred blocks from the epicenter of the economic and political Buenos Aires (CABA) activities. In order to evaluate and quantify the variations of the noise level in the involved area, a strategic noise map - that includes the edificial and vehicle conditions before and after implementation of the project - has been made. Cadna-A software has been used for its development. This paper describes the process of the reached acoustic models, displaying their results. It is concluded that although there was a decrease of vehicular noise in much of the affected area there were certain parts of deterioration, that should be reevaluated. Even with these improvement, redistribution of noise levels in the sector continues, in most cases, exceeding the recommendations and limits established by current legislation.

Wednesday, 7 September 2016

Wednesday morning, 7 September 2016 **Microcinema**
09:00 - 10:40
FIA-AA - Architectural Acoustics - Room and Building Acoustics
FIA-AA - Acústica Arquitectónica - Acústica de Salas
FIA-AA - Acústica Arquitetónica - Acústica de Salas

Architectural Acoustics - Room and Building Acoustics:
Paper FIA2016-52

Noise map as support instrument to achieve acoustic performance of residences envelopment according to NBR 15.575-2013

Fabiana Curado Coelho[a], Cândida Maciel[b], Jhennyfer Loyane Gama Pires[c], Ludmila de Araujo Correia[d], Luis Fernando Hermida Cadena[e]
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Abstract
Since 2013, the ABNT NBR 15.575-2013: Edificações Habitacionais – Desempenho defined performance parameters of residential buildings, among which are inserted acoustic items. This standard interferes in the projection methodologies of new buildings, including the definition of envelopment. The noise exposure analysis of housing is required to fit it into one of the three noise classes defined by the standard, in order to determine the minimum performance of air sound insulation in every situation. The noise mapping is one of the instruments for the determination of these classes, adopted by its distinguished ability to represent the sound levels, as well as simulation of future scenarios. The aim of this study was to assess the relevance of this instrument for the classification of a residential building, as well as its influence on the definition of constructive guidance throughout the project stages. For case study adopts a building in a satellite town of the Distrito Federal. The method began with the development of the noise mapping in the evaluation and prediction software of environmental noise. As input data were used cadastral maps, number of vehicles, legal project of the building venture, existing sound sources and sound pressure level measurements. The obtained results identified that the in-depth study of the sound environment of a site can bring more efficient alternatives in decision-making at different stages of the project, especially if consolidated from the early stages. Therefore, it enables a higher acoustic quality of the project, besides the reduction in the cost required to adapt the acoustic performance.

Architectural Acoustics - Room and Building Acoustics:
Paper FIA2016-72

Acoustic quality in shopping malls for environmental certification. Case studies: Shopping malls RioMar Recife (PE) and Fortaleza (CE)

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Abstract
Designing a Shopping Mall is a complex activity involving a multidisciplinary team in order to find compatible solutions. Much is invested in physical infrastructure, but special attention should be given to the acoustic quality of mall spaces. One of the advantages of acoustic comfort is that it may prolong

customers stay in these places, and increases the probability of consumption. So, acoustic comfort is not a cost, but an investment. One of the requirements of the Certification Process AQUA-HQE (Brazil's version of the French Démarche Haute Qualité Environmentale) for the environmental quality of malls is the control of internal acoustics in common customer hall spaces by performing an specific study involving reverberation time control and the background noise level reduction to provide listening comfort and satisfactory communication to customers. This study was conducted during the Design Phase of AQUA-HQE in two Shopping Malls in Brazil, one in the city of Recife (Pernambuco) and the other in Fortaleza (Ceará). The simulations were developed in the software EASE 4.4 (Enhanced Acoustic Simulator for Engineers) and with spreadsheets of "Noise Reduction Level" and "Signal-Noise Ratio" in Microsoft Excel. The studies predict the noise level in case of no acoustic treatment and with proper acoustic solutions. Then, during the Realization Phase, after the inauguration and regular operation of the malls, internal measurements of sound pressure levels were performed to compare and validate the noise level values previously estimated. The results obtained in field validated the methodology used at the design process and helped to understand sound behavior, which allowed recommendations for further studies and projects.

Architectural Acoustics - Room and Building Acoustics:
Paper FIA2016-93

Case study of acoustic performance of corrections in junctions of internal wall and curtain wall façade

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Abstract
A building is composed of several subsystems, each one with many specifications for which the building project compatibilization is an essential tool since the initial phase of the design. However, in some cases, not always the building project compatibilization is made and the acoustic insulation failures of the enclosures can be indicated as one of many others undesirable consequences. This paper addresses the results of airborne sound insulation of corrections made in the junction between the internal masonry walls and the curtain wall façade. Three correction proposals for the failure resultant from the incompatibility between the internal walls and the frame modulation of the curtain wall were executed in an office building. The field measurement were applied according to the ISO 16283-1 and the results obtained show an increase in the D'nT,w between 8 and 9 dB.

Architectural Acoustics - Room and Building Acoustics:
Paper FIA2016-51

Intervenients in floating floors execution process and its reflects on acoustic performance

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Abstract
With all the demands taken place after the implementation of ABNT NBR 15.575/2013, the Brazil undergoes a new stage on the execution of residential buildings. However, it cannot be always perceived that the labour force has adapted to this new reality and, on what concerns acoustic aspects, those difficulties are fairly evident. The lack of qualification of the labour force in an acoustic solution can interfere directly in the quality of specific systems, as it is clearly observed in the noise attenuation of flooring with disconnection overlays. With the aim to identify and assess the intervenient variables in the process of the execution of flooring with disconnection overlays, a case study was undertaken in five construction sites, with the assembling of three different flooring systems. Subsequently, it was measured the weighted noise pressure levels of standardized impact for each of the systems built, according to the norms procedures ISO 140-7 and 717-2. The obtained outcome was compared to the performance expectations released by the overlays producers. Discrepancy was

noticed in the performance of the same overlay installed in different construction sites; also many outcomes did not reach the expected result as their producers'. The detailed analysis of the overlay installation process allowed the identification of labour flaws as the main source of attainment loss. It was also observed that the executive details were not adequately followed, due to the procedures adopted on the constructive process management. The work undertaken shows the importance of labour training and the attention to the constructive process stages to reach the desired acoustic outcomes, therefore contributing to the improvement of the flooring systems installation.

Architectural Acoustics - Room and Building Acoustics:
Paper FIA2016-67

Influence of the compression conditions in the acoustic performance of resilient layers of floors

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Abstract

The acoustic performance of floor systems is directly linked to the specific characteristics of each material, but the behavior of these materials can be changed along time with the common use of the building. Therefore, the reduction of the resilient layer thickness of the floating floors represents a loss in efficiency of the acoustic system, which can be caused either by compressions along the life cycle of the building or the compressive loads resulting from accidental loads. These conditions may indicate that a same product can sometimes present different performances due to the different compositions of floating floor systems. This paper presents a study based on five different underlayers of floating floors. The underlayers were polymeric fibrous materials with densities between 180 and 1000 kg/m² which wasevaluated before and after compression 122 days.

Wednesday morning, 7 September 2016 **Microcinema**
11:00 - 11:20
FIA-AA - Architectural Acoustics - Room and Building Acoustics
FIA-AA - Acústica Arquitectónica - Acústica de Salas
FIA-AA - Acústica Arquitetónica - Acústica de Salas

Architectural Acoustics - Room and Building Acoustics:
Paper FIA2016-45

Study of sound absorption of wools composed by PET waste products

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Abstract

The pursuit for recycled materials is in consistent extension, since the environmental damage caused by not biodegradable materials need extremely concern when discarded in an incorrect way. The high period needed those materials to degrade in a spontaneous way or the outgassing descendant of the burn process when they are incinerate. Among the diversity of waste materials, we meet the PET (Polyethylene terephthalate) waste, used in large scale in the worldwide. The reuse of those materials are of extreme significance for the circularity of their proposes. Since this, a wool for use in sound absorption proposes in room acoustics was developed using PET waste. The purpose of this article is to investigate the use of these PET wools in different thickness and densities, analyzing their sound absorption capacity and comparing the different compositions of wools in a manner to determine the highest sound absorption composition. For this, the test method used was described by ISO 354:2003, that determine the mounting types for this type of material as well as the sound absorption test procedure, as well as ISO 11654:1997 to calculate the weighted sound absorption coefficient. For the tests, the wools were placed directly to the floor of the reverberant room as well as placed with an air coating of 5 cm underneath them. The results had shown that the wools with more thickness and density had a better sound absorption coefficient compared to the others, so as the mounting type has significant gains and losses to the sound absorption capability of every PET wool tested.

Wednesday morning, 7 September 2016 **Microcinema**
11:20 - 12:00
FIA-SS - Soundscape
FIA-SS - Paisaje sonoro
FIA-SS - Paisagem sonora

Soundscape:
Paper FIA2016-122

Learning the soundscape in urban and architectural itinerary: Listening Barcelona blindfolded

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Abstract
The itinerary proposed to be listened is in fact one of the most suitable methods to understand the sound of the city. We have prove that learning the soundscape improves extraordibarily when the space is wandered blindfolded, especially if the subject is the same citizen. For this reason, ten paths through Barcelona have been proposed, one for each district in wich the city is divided. Five of them have already been done through the intervention of “La Fàbrica del Sol”, a municipal entity. The work has been arrange by “Àrea de Medi Ambient” of Barcelona City Council.

Soundscape:
Paper FIA2016-111

Some consideration on the design of listening-tests for soundscape assessment

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Abstract
The assessment of soundscape perception is often addressed through listening-tests. Notwithstanding some limitations, such as the reduced range of conditions that can be studied as well as the extrapolation of the obtained results to real-life scenarios, several authors have used this technique for different approaches and objectives due to its ability for controlling research parameters, which would be very difficult or impossible to control in in-situ evaluations. This work is aimed at conducting a non-exhaustive review of those factors that must be considered in the design stage of a listening-test. Based on the experience previously acquired by this research group, but also from a literature review, some recommendations are provided (and discussed) for the appropriate design of a listening-test. These recommendations are intended to support the researcher in adjusting the listening-test design to the objective of the research, therefore ensuring a much more appropriate soundscape assessment.

Signal Processing in Acoustics:

Paper FIA2016-35

Simplified filter bank to emulate de head diffraction depending on the azimuth angle of the source

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Abstract

Amplitude panning is the most common panning technique. Another method is time panning with a constant delay applied to one channel in stereophonic listening. Time panning is typically not used in stereo imaging, but it can be used when some special effects are created. The maximum interaural difference in time of arrival of air propagated signal is about 700 µs. Binaural hearing with headphones using that delay is not perceived as a virtual source at 90º of azimuth angle. However, including spectral modifications created by head diffraction, the azimuth angle can be perceived as it is expected. A simplified head diffraction model with possible applications in audio mixing is presented. The model is based on a bank of filters to emulate the angular position of the source in the horizontal plane. The aim of the model is to emulate a virtual position of the sound source with minimum computational effort when compared to the convolution with the corresponding head related impulse responses. Our results suggest that diffraction loss can be adequately represented by shelving filters. The CIPIC public database of head related impulse responses was used to compute the parameters of the simplified filter bank. Individuals with identical head diameter are selected from the database and their impulse responses are convolved with a series of one-third octave band noise in order to obtain for each ear a power profile in terms of the frequency. The results are averaged in order to smoothen the individual differences, thus neglecting other spectrum disturbances that do not correspond to diffraction. The model provides us with parameters about frequencies and attenuation levels of the shelving filters..

Signal Processing in Acoustics:

Paper FIA2016-53

Improving the estimated acoustic absorption curves in impedance tubes by using wavelet-based denoising methods

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Abstract

The two-microphone impedance tube method is widely used to obtain experimental sound absorption curves of test specimens of acoustic materials and absorbers. The measurement noise is a major issue in impedance tube experiments because the quality of the estimated absorption curves degrades quickly as the signal-to-noise ratio (SNR) decreases. The SNR is reduced at low frequencies due to the poor loudspeaker response and to the fact that the spacing between microphones becomes a fraction of the acoustic wavelength. This paper makes use of wavelet-based denoising methods to increase the SNR in impedance tubes through digital signal processing. The two sound pressure signals that would be sensed in a real experiment are simulated by a time-domain numerical procedure based on the finite element method. An exponential sine sweep is used as the loudspeaker excitation signal. The simulated microphone signals are then polluted with additive white Gaussian noise to mimic the measurement noise. Next, this noise is partially removed through wavelet denoising. Finally, the filtered simulated signals are subjected to the same signal processing

procedure that would be applied to real measured signals to derive the absorption curve, which consists basically in a frequency response estimation by using the discrete Fourier transform. Numerical results are reported for a simple test specimen with a known absorption curve, namely, a loudspeaker acting as an electroacoustic absorber. The performances of different wavelet families are investigated. The comparison of the “exact” absorption curve with the estimated ones with and without noise removal shows that wavelet techniques lead to improvements in the impedance tube results, especially at low frequencies.

Wednesday afternoon, 7 September 2016 **Microcinema**
15:10 - 16:10
FIA-MU - Musical Acoustics
FIA-MU - Acústica musical
FIA-MU - Acústica musical

Musical Acoustics:
Paper FIA2016-22

Ultrasonic components of musical instruments

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Abstract
There are few works caracterizing the ultrasonic components emitted by musical instruments, primarily because it did not seem worthwhile to measure an energy that was difficult to register and that it is not perceived by humans. Register sounds over 20 kHz are possible today with typical audio equipment. There exists low cost equipment such as microphones, sound cards and speakers that work with frequencies extending the traditional HiFi standard. The possibility of perceiving frequency components over the audible range is still a matter of debate, but we think that it is important to know how much energy there is in musical instruments in the ultrasonic range in order to fuel that debate. In a previous work, energy measurements over the standard audible range have been made for some musical instruments by our research group. This paper reports new findings about the ratio of high to low frequency energy over time for several musical instruments and different angles.

Musical Acoustics:
Paper FIA2016-96

The percussion as acoustic element in orchestras and musical groups from the twentieth century

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Abstract
Mathematical and also physical possibilities involving the study and development of percussion are numerous. The percussion study gained new momentum from the XX century, with composers such as Stravinsky, Shostakovich, Carl Orff and others. The idiophones and membranophones, which are part of the group of percussion instruments, are highly versatile and different from traditional orchestral instruments. Some have a definite pitch, like the xylophone, the marimba, the vibraphone and timpani. Among the indefinite sound we can mention the tomtom, the snare drum, the bass drum and cymbals. We did a physical analysis of some percussion instruments: xylophone, tambourine, snare drum, tom-tom and triangle. We analyze the sound pressure levels, the decays and sound spectra using different attacks, and in the case of xylophone, different notes. We note the diversity of behaviors and effects possibilities.

Musical Acoustics:

Paper FIA2016-115

Musical imagery: Influence of musical training in auditory sound recognition by melodic and rhythmic modified stimuli of long latency AEPs

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Abstract

The purpose of music is communication, so that musical structure has to take into account the structure of memory. Studies in neuroscience demonstrate two cognitive paradigms, "classical" with serial processing of information, and “non-classical” with parallel processing. Processing in memory structure links both paradigms and is schematically divided into three stages: immediate, short term and long term memory. This study explores short and long term memory through the stimulation with rhythmic and melodic patterns, highlighting the difference in neurological processing of people with musical training. The subjects studied are divided into 2 groups: first 12 men and women with studies and practice with musical instruments not less than 5 years, the other group comprehends 12 men and women without any formal studies in music and that do not play any musical instrument. Both groups first will be tested with SLAEPs (short-latency auditory evoked potentials) to verify a good ability to listen, then will be evaluated under a protocol of long latency AEPs with a stimuli of 4 groups of 9 notes artificially created: a melody with a simple rhythm, a simple melody with a variable rhythm, a simple rhythm with varying melody and a variable rhythm and melody. This evaluation will analyze short-term memory by the exposure to new stimuli for subjects (learned music). The same group of subjects was then exposed to known musical stimuli, to evaluate both the long-term memory and imagination. The data obtained in each test will be analyzed by musical knowledge and sex. The results allow evaluating quantitative and/or qualitative differences between groups of subjects based on musical experience and acquired underlying differences in brain anatomy involved in musical memory.

Thursday, 8 September 2016

Thursday morning, 8 September 2016 **Lounge Lateral Room**
09:40 - 10:00
POSTER SESSION - Monitor 3
FIA-PP - Noise: Sources and Control
FIA-PP - Ruido: Fuentes y su Control
FIA-PP - Ruído: Fontes e o seu Controlo

POSTER
Noise: Sources and Control:
Paper FIA2016-95

Sound analysis of Copacabana neighborhood traffic routes, Rio de Janeiro

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Abstract
With the growth of cities, noise pollution has become a major environmental problem. While vehicle traffic is a major source of urban noise, the study of solutions for mitigation should also consider the characteristics of urban form (width of roads / height of buildings, topography, land cover, urban mesh density). In summary, the solutions to reduce noise pollution undergo solutions within the planning and management of land use and urban mobility. Taking into account these factors, it is necessary to further deepening the study of the road as a source of noise and its relation to the built volumes. The acoustic classification of traffic routes, as practiced in EU countries, is an important element in this context. This work - carried out in a number of roads in the city of Rio de Janeiro - presents the acoustic analyzes conducted on a set of Copacabana neighborhood of roads in the city of Rio de Janeiro, trying to relate the functional pathways classification (expressed, arterial, collector and local) and its morphological characteristics, with sound pressure levels emitted. As analysis methodology NPS and simulationsmeasurements were performed in SoundPlan program.

POSTER

Psychological and Physiological Acoustics:

Paper FIA2016-79

Environmental noise of schools and its relation with cognitive performance and cortisol levels in adolescents

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Abstract

In school environment, noise produced inside and outside the classrooms can impair student´s cognitive performance and health. Thus, the present study aimed to relate the environmental noise of schoolswith cognitive performance and cortisol levels in adolescents. Two public schools from a city in south Brazil were evaluated, one does not have heavy urban traffic in its surroundings (school A) and the other one has it (school B). The students of the two schools have similar characteristics. To evaluate cognitive performance, concentration test was carried out. Cortisol levels were evaluated in the beginning and at the end of the class, in the morning shift. The comparison of continuous variables was performed through independent-sample T test. Values of$p<0.05$ were considered significant. The results demonstrate that adolescents from school B presented lower mean for the concentration test(86 versus 99 points; $p=0.033$). In addition, cortisol levels in adolescents from school B decrease less($\Delta=0.208$ µl/dL), during the morning, in comparison to the students from school A($\Delta=0.106$ µl/dL) ($p=0.017$). We conclude that adolescents from the school with the worst acoustic performance present more unsatisfactory results for cognitive performance evaluation and cortisol levels.

POSTER

Soundscape:

Paper FIA2016-60

Listening to faith: Description of the allotment Village Campestre soundscape, in Maceió, Brazil

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Abstract

The sounds produced by churches have psycho spiritual values. The celebrations are taped for respect and admiration from the faithful people. Due to this space´s acoustic inadequacy, the religious sounds invades the public space and reach the neihborhood, and may cause discomfort. To evaluate these sounds with acoustic satisfaction, it is necessary the adoption of qualitative parameters, so its insertion in the local soundscape may be understood. In front of the great variety of cognizable sounds in the urban environment, the sound evaluation methods within the rules and laws are not satisfactory. These methods don´t use qualitative analysis, making it a disadvantage to the sound subjectivity reasoning. In this regard, this work aims to characterize the allotment Village Campestre soundscape, and verify the relevance of the sounds produced by churches in this scenario. In order to accomplish this study, a mixed method of case study qualitative and quantitative evaluation was adopted. It has been carried out the study object description, acoustic data collection according to Brazilian rules, qualitative description of the identified sound events, and in the end, a form based interview about the sound perception in the area that was carried out with the residents. With the results analysis, it was noticed that the quantitative acoustic data broke through Brazilian rules limits, with high sound levels that may cause disconfort. However the qualitative data proved that the psycho spiritual factor interferes with a sound interpretation. As to the local soundscape description, it was noticed the insertion of religious sounds, the lack of concern regarding the quality of sounds produced in the city and the low risk to cause sensations on the individuals.

POSTER

Noise: Sources and Control:

Paper FIA2016-64

Acoustic performance to airborne noise in vertical partitions on construction light steel framing. Case study: Residential building, Pernambuco- Brazil

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Abstract

Due to the rapid and unplanned growth of cities, and the lack of effective urban planning, the company started to deal with serious environmental problems that directly affect the population's quality of life, affecting among other things the sound quality of spaces. Nowadays, the construction industry is envisioning the introduction of more effective systems from an environmental point of view. In this context, a widely applied technology is the Light Steel Framing (LSF), whose marketing propagates an efficient acoustic performance of the building. This study aimed to evaluate the acoustic performance of the Internal and External Vertical Seals System (SVVIE) in a single-family residential building in the city of Vitoria do Santo Antão, state of Pernambuco- Brazil, made with constructive system LSF in order to determine data that can contribute to application parameters in construction. The method adopted included the achievement of on-site measurements, according to the parameters set out in ISO 16283:2013 and ISO 3382: 1997. The data obtained in the measurements was investigated with the help of Software Solo 01 dB Bati 32 in order to check the sound insulation potential of the experiment, relative to airborne noise, by obtaining the standardized difference weighted level two meters of facade (D_2m,nT,w) and standardized weighted difference level between environments (DnT,w) from the graphing (spectrum frequency x insulation) instead of the ISO settings 717-1:1996. And finally proceeded with a comparative analysis of the results achieved with the sound insulation requirements stated in NBR 15575-4:2013.

POSTER

Noise: Sources and Control:

Paper FIA2016-17

Sound power comparison between a new device and a daily use device under the standard ISO 3741:2010

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Abstract

Despite the sound power specifications under the standard expressed by the manufacturers in their products, it is necessary to know how long was used the product before to perform this type of measurement. This is because the deterioration of time could increase the noise of the machine. The aim in this research is to disseminate the sound power measurement results of an automatic washing floor machine using the standard ISO 3741:2010. The research does a detailed comparison between the results obtained in the measurement and the specifications presented by the manufacturer. The investigation ends with the characterization of the noise produced by the engine and concludes by suggesting supplements in the measurement under the standard.

POSTER

Noise: Sources and Control:

Paper FIA2016-27

Noise impact assessment around bus terminal. Case study: The bus terminal of Cruz das Almas - Maceió/AL

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Abstract

The urban noise is one of the biggest problems that people experience nowadays, directly affectinginner public areas and partsof buildings too. The most responsible for that are vehicles, such as cars of various sizes, motorcycles, trucks, buses and others. The existence of bus terminals in neighborhoods is very important, in order to facilitate urban mobility enabling a large number of people to reach their destinations using the same means of transportation, reducing traffic, accidents and environmental pollution. According to studies, public transportation is considered accessible when the station is located at least 400 meters from buildings. However, in most cases, bus conglomerates cause higher noise than what is considered acceptable by the rules, causing discomfort, resulting instress, illnessand other disorders directly affecting people. The aim of this study is to evaluate the impact of noise in the surrounding areas of a bus terminal, located in the neighborhood of Cruz das Almas, Maceió/AL, Brazil and how noise interferes in the environment, which was observed from a square. Levels of noises were measured in the morning and afternoon in the surrounding areas of this bus terminal. The experience was made considering four outdoorand, at differents times in the day. Results showed that the outside environment Laeq, reached high levels, all above what was recommended by the standard rules and also the square and nearby buildings were affected. Thus, the location of bus terminals need to be carefully chosen in order not to raise problems related to acoustic impact.

Thursday afternoon, 8 September 2016 **Lounge Lateral Room**
14:50 - 16:10
POSTER SESSION - Monitor 3
FIA-EN - Environmental Acoustics & Community Noise
FIA-EN - Acústica Ambiental y Ruido Comunitario
FIA-EN - Acústica Ambiental e Ruído Comunitário

POSTER

Environmental Acoustics & Community Noise:

Paper FIA2016-30

Evaluation of acoustic comfort in long stay institutions for elderly in Maceio-AL

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Abstract

Current knowledge of Science added to advances in political and technical systems in health have passed to the intense process of population aging in the country. The elderly person, over 60 years, suffers from the progressive degeneration of the body, becoming a victim of discrimination, neglect and maltreatment, many are sent to Long institutions Permanence Elderly (LTCF). The issue that goes unnoticed are the noise levels in these institutions. This Article is justified by the need for acoustic comfort for providing the quality of life of the elderly. The objective of this study is to evaluate the acoustics of permanence environments of a nursing home for elderly punctuating environments into three categories: healthy, tolerable and / or unhealthy. The methodology used consisted of a literature review through these types of environments where they were made technical evaluations of selected shelters through the ambient noise level, noise ratio and signal intelligibility test application with the elderly with better understanding. Data analysis was guided by the NBR 10151: 2000 and NBR 10152: 1997. The conclusion of the study established that in ILPIs need special care because their users have different mental states, lucid and not lucid (sclerotic), where each has different needs. It was observed that the "lucid" elderly require more careful acoustic environments. In the assessments should highlight the care with dormitories and environments that require a better acoustic comfort. In conclusion, it can be seen that these environments for the elderly (LTCF) deserve greater care when the occupational assessment in the pursuit of sound quality.

POSTER

Environmental Acoustics & Community Noise:

Paper FIA2016-34

Annoyance of the traffic noises in the city of Buenos Aires

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Abstract

A sound event can be determined and fully characterized by its spectral and temporal behavior. But it is more complex to determine the degree of annoyance of the same event because it depends on subjective aspects of each individual. This study investigates the annoyance associated to eight traffic noise events of cars, buses, motorbike. First, traffic noises were recorded in the Autonomous City of Buenos Aires (CABA) in three streets with different traffic flow characteristics. Then, a paired comparison test was performed to determine the degree of annoyance associated to these stimuli. Scale values of annoyance were obtained and correlated with objective parameters and sound descriptors commonly used: equivalent continuous sound pressure level (L_{eq}) with and without weighting, peak level (L_{peak}), sound exposure level (SEL), crest factor, duration and normalized octave frequency band level. It was found that the highest correlation with the degree of annoyance were the L_{Aeq}, the SEL and the spectral content of medium and high frequencies (2-8 kHz range).

POSTER

Environmental Acoustics & Community Noise:

Paper FIA2016-61

Evaluation of the impact of tall buildings construction in the acoustic quality of indoor environments and urban grid of the neighborhood

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Abstract

For the expansion of cities the implementation of infrastructure and services is required, leading to increased construction works, which in the center of the concept, they are emitters of noise polluters in the environment. Under these circumstances, it is extremely important to monitor and evaluate the urban environment since it directly influences the general state of health.This study aims to analyze the sound impact generated by the noise construction site of a multifamily building, checking how the interaction between sound sources of the work and the urban form influences in particular soundscape, with a case of study if a portion of the urban area located in the neighborhood of Jatiúca, in the city of Maceió, Alagoas, Brazil. It was collected physical data and measurements of sound pressure level of the urban area with the active and inactive construction. The study was based on the application of a software for mapping and acoustic prediction which was analyzed, first, the current sound setting and then the hypothetical scenarios created. The results in measurements in situ under the influence of active work were above the permitted by applicable Brazilian standard, NBR 10151

(ABNT, 2000) and the simulations showed that the physical traits of urban form influence the sound propagation outdoors and, therefore, the sound environment of a particular region.

POSTER

Environmental Acoustics & Community Noise:

Paper FIA2016-62

Night noise impact caused by commercial establishments: Case study avenue Dr. Antonio Gomes de Barros, Maceió-AL

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Abstract

The disorderly growth of cities result in problems on vehicular traffic and uses and occupations of buildings, factors related to urban noise. In the absence of environmental supervision, the buildings are modified and become increasingly trades of evening entertainment, without previous studying of the impacts caused to the region by them and by the generated traffic. Jatiúca district is one of three major targets of complaints registered by the Municipal Department of Environmental Protection, which the majority of cases are related to the noise pollution. Among the records, bars and restaurants are as the two major causes of noise discomfort. Thus, this study aims to evaluate the noise impact caused by night establishments in a residential predominance area and divided into 29 evaluation points. The methodology applied included data collection of: uses and occupations, photography, sound pressure level, vehicle light and heavy traffic, and night commercial establishments contained in the region with days and hours of operation and characterization of sound instruments used for the public's entertainment. In data analysis raised was found sound pressure level values above 50dBAthan is recommended by the standard. This value is double the indicated to the region, worrying factor for the health of the community. The high rates found come from commercial fixed sources and engine traffic or from the inappropriate use of speakers coupled in vehicles. So, all make us take the knowledge that the inspection on the acoustics of the cities should be done regularly and not only through community complaints, since the source causing noise can be mobile and individual.

Thursday afternoon, 8 September 2016 **Lounge Lateral Room**
16:30 - 17:30
POSTER SESSION - Monitor 3
FIA-AA - Architectural Acoustics-Room and Building Acoustics
FIA-AA - Acústica Arquitectónica - Acústica de Salas
FIA-AA - Acústica Arquitetónica - Acústica de Salas

POSTER

Architectural Acoustics - Room and Building Acoustics:

Paper FIA2016-56

Evaluation of the acoustic performance of systems of walls and floors for verifying compliance with standard 15.575

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Abstract

The present study aims to evaluate the acoustic performance of vertical and horizontal fencing for residential buildings in order to verify the compliance with the performance standard-NBR 15,575 (2013). The methodology of work is based on a data collection where we sought to identify what are the constructive systems most often used in the city of Maceió-AL. From the results obtained, ten buildings were selected with disparate systems to be evaluated. External vertical seals of dormitories were evaluated, internal vertical seals between apartments and horizontal seals of the impact noise and air. From the results obtained, one can verify on which levels of the buildings were inserted. The results reveal a worrying factor: most evaluated facades have acoustic insulation levels lower than the standard sets a minimum and mandatory. Regarding types of internal walls, a large variation was found the results even when dealing with similar construction systems. The results of flooring systems were more satisfactory levels therefore, it has levels that fall within the minimum performance parameter specified by the standard.

POSTER

Architectural Acoustics - Room and Building Acoustics:

Paper FIA2016-76

Results comparison of acoustic performance of a church with contemporary architecture measured with two types of impulsive source

Marselle Nunes Barbo[a], Dinara Xavier da Paixão[b], Leonardo Arzeno[c], Moisés Canabarro[d]

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Abstract

The paper presents a results comparison to analyze acoustic performance in a church with contemporary architectural style, using measurements with two different models of impulse source. This particular religious building differs from traditional churches in that its shape is that of a bell. The church can hold up to nine hundred people seated and can accommodate an additional nine hundred people standing occupants using available open floor areas. The physical space is characterized by reflective materials such as concrete and glass, placed together in unique custom designed panels that resemble mosaics. The seats are wooden and floors are covered in ceramic tiles. The values of Reverberation Time (RT60), Early Decay Time (EDT), Clarity (C80) and Definition (D50) were measured and will be presented in the paper. The evaluation of the acoustic quality parameters for the surroundings was done with manual impulse alternative source and omnidirectional impulse source. The tests enabled the results comparison. The information contained in this research, supported by references and experiments, constitute a contribution towards finding alternative impulse sources, when omnidirectional impulse source is not an option due to its high cost and difficult handling.

POSTER

Architectural Acoustics - Room and Building Acoustics:

Paper FIA2016-70

Uncertainty in measurement determination in the sound reduction index in laboratory

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Abstract

The determination of measurement uncertainty is an evaluation parameter of the regularity and reliability of the test results performed by the laboratory and the Guide to the Expression of Uncertainty in Measurement, ISO/IEC Guide 98-3:2008, is the reference standard used to determine the measurement uncertainty. All achieved magnitude in a test must be followed by its uncertainty estimation, thereby providing a representation of the laboratory technical knowledge of the possible factors contributing to the random error of this magnitude. As to the building acoustics, the ISO 12999-1 specifies procedures to the evaluation of measurement uncertainty, as well as maximum values of standard deviation to be followed and the standard uncertainty to be used when there is no possibility of performing inter-laboratory tests. This study aims to evaluate the components that influence the

result of the sound reduction index of a vertical sealing system tested in the laboratory, according to the procedures of ISO 10140-2:2010.

POSTER

Architectural Acoustics - Room and Building Acoustics:

Paper FIA2016-38

Sound insulation field measurements of Brazilian traditional internal horizontal partition systems

Josiane Reschke Pires[a], Sérgio Klippel Filho[b], Rafael Trevisan[c], Maria Fernanda de Oliveira Nunes[d]

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Abstract

Sound insulation field measurements allow the evaluation the conformity of the rooms analyzed according to the requirements regulated by current standards. The internal horizontal partitions with hollow, traditionally used in Brazilian residential buildings do not meet the higher-grade levels of the Brazilian performance standard. The aim of this paper is to analyze the acoustic insulation of internal horizontal partition systems of a residential building consisting of different types of building systems and coatings. Assays were carried out in the field, following the procedures prescribed in ISO 16283-1:2014 in drywall, masonry concrete hollow bricks and two kinds of masonry ceramic hollow bricks filled with sand, with different coating composition. The results showed that the walls of masonry ceramic hollow bricks with alveolus filled with sand presented superior acoustic insulation to other systems analyzed.

POSTER

Architectural Acoustics - Acoustic Rooms:

Paper FIA2016-46

Characterization of resilient material used in floating floor systems

Letícia Zuchetto[a], Rafael Trevisan[b], Maria Fernanda Nunes[c]

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Abstract

The construction market in Brazil has been adapting to qualify residential buildings in the country. One of these improvements is the impact noise attenuation, provided by inserting a resilient material between the slab and the subfloor. Therefore, it is important that these insulating materials are properly characterized, so that designers can indicate the appropriate materials for each building. Thus, this paper, proposes to characterize nine types of resilient materials currently available in the Brazilian market through dynamic stiffness testing, according to ISO 9052, and insulation to impact noise, according to ISO 10140-3. The study also compares values of insulation to impact noise obtained in the laboratory with the values provided by EN 12354 through dynamic stiffness and subfloor mass values. The results obtained ranged from 8 to 27dB of weighted sound reduction in the level of impact pressure.

POSTER

Architectural Acoustics - Room and Building Acoustics:

[Paper FIA2016-55]

Sound insulation comparative study on ceramic masonry with mortar or plasterboard coating

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Abstract

Within the evolution of civil construction in recent years, many researchers have been arranged focused in buildings performance, in order to develop materials and methods used in this branch. The study of materials and its properties aims to make construction economical, efficient, durable and providing comfort for its users. Among buildings characteristics, acoustical performance is significantly affected for materials choice. This research aimed to compare acoustical performance of masonry walls coated with mortar or plasterboard. To accomplish this objective, two surface densities of plasterboard (8,5 kg/m² and 12 kg/m²) were tested in various compositions: pasted boards, boards fixed in metalic profiles and combined systems. In this composition fixed in metalic guides. The types set on metal guides also assessed whether to influence the use or not of glass wool in the gap between the profiles and the substrate. The results show that in similar systems, the use of glass wool in filling the gap between the plate and the substrate provided better sound insulation. It was also noted that the higher density boards and joint systems provide superior results than the plates of lower density and systems without change of form of installation.

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