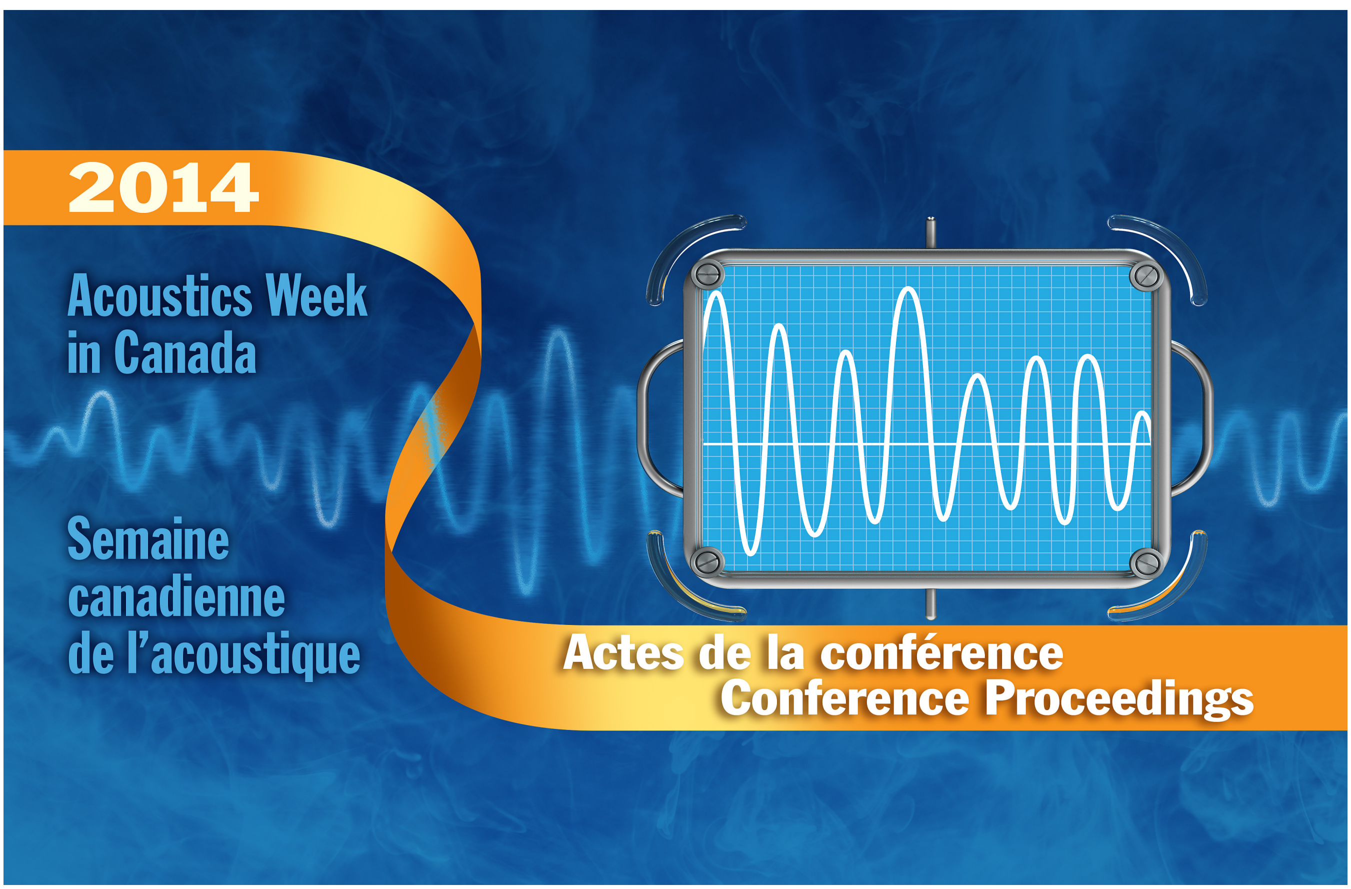
ISSN 0711-6659

canadian acoustics
acoustique canadienne

Journal of the Canadian Acoustical Association - Revue de l'Association canadienne d'acoustique

SEPTEMBER 2014 | SEPTEMBRE 2014
Volume 42 - - Number 3 | Volume 42 - - Numéro 3



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THE CANADIAN ACOUSTICAL ASSOCIATION P.O. BOX 1351, STATION "F" TORONTO, ONTARIO M4Y 2V9

L'ASSOCIATION CANADIENNE D'ACOUSTIQUE C.P. 1351, SUCCURSALE "F" TORONTO, ONTARIO M4Y 2V9

Canadian Acoustics publishes refereed articles and news items on all aspects of acoustics and vibration. Articles reporting new research or applications, as well as review or tutorial papers and shorter technical notes are welcomed, in English or in French. Submissions should be sent only through the journal online submission system. Complete instructions to authors concerning the required "camera-ready" manuscript are provided within the journal online submission system.

L'Acoustique Canadienne publie des articles arbitrés et des informations sur tous les aspects de l'acoustique et des vibrations. Les informations portent sur la recherche, les ouvrages sous forme de revues, les nouvelles, l'emploi, les nouveaux produits, les activités, etc. Des articles concernant des résultats inédits ou des applications ainsi que les articles de synthèse ou d'initiation, en français ou en anglais, sont les bienvenus.

Canadian Acoustics is published four times a year - in March, June, September and December. This quarterly journal is free to individual members of the Canadian Acoustical Association (CAA) and institutional subscribers. Canadian Acoustics publishes refereed articles and news items on all aspects of acoustics and vibration. It also includes information on research, reviews, news, employment, new products, activities, discussions, etc. Papers reporting new results and applications, as well as review or tutorial papers and shorter research notes are welcomed, in English or in French.The Canadian Acoustical Association selected Paypal as its preferred system for the online payment of your subscription fees. Paypal supports a wide range of payment methods (Visa, Mastercard, Amex, Bank account, etc.) and does not requires you to have already an account with them. If you still want to proceed with a manual payment of your subscription fee, please complete the application form and send it along with your cheque or money order to the secretary of the Association (see address above). - Canadian Acoustical Association/Association Canadienne d'Acoustique P.B. 74068 Ottawa, Ontario K1M 2H9 Canada - (613) 562-5248 - (613) 562-5800 p. 3066 - secretary@caa-aca.ca - Prof. Chantal Laroche

Acoustique canadienne est publié quantre fois par an, en mars, juin, septembre et décembre. Cette revue trimestrielle est envoyée gratuitement aux membres individuels de l'Association canadienne d'acoustique (ACA) et aux abonnés institutionnels. L'Acoustique canadienne publie des articles arbitrés et des rubriques sur tous les aspects de l'acoustique et des vibrations. Ceci comprend la recherche, les recensions des travaux, les nouvelles, les offres d'emploi, les nouveaux produits, les activités, etc. Les articles concernant les résultats inédits ou les applications de l'acoustique ainsi que les articles de synthèse, les tutoriels et les exposées techniques, en français ou en anglais, sont les bienvenus.L'Association canadienne d'acoustique a sélectionné Paypal comme solution pratique pour le paiement en ligne de vos frais d'abonnement. Paypal prend en charge un large éventail de méthodes de paiement (Visa, Mastercard, Amex, compte bancaire, etc) et ne nécessite pas que vous ayez déjà un compte avec eux. Si vous désirez procéder à un paiement par chèque de votre abonnement, on vous invite à compléter le formulaire d'adhésion et l'envoyer avec votre chèque ou mandat au secrétaire de l'association (voir adresse ci-dessus). - Canadian Acoustical Association/Association Canadienne d'Acoustique P.B. 74068 Ottawa, Ontario K1M 2H9 Canada - (613) 562-5248 - (613) 562-5800 p. 3066 - secretary@caa-aca.ca - Prof. Chantal Laroche

Welcome To Winnipeg

ACOUSTICS WEEK IN CANADA

October 8-10, 2014

Assiniboine River Walk
Photo by Ruehle Design, Courtesy Tourism Winnipeg

The Forks
Photo by Dan Harper, Courtesy Tourism Winnipeg

Organizing Committee

Conference Chair:	Karen Turner, Protec Hearing Inc.
Technical Chair:	Ramani Ramakrihnan, Ryerson University
Exhibit Coordinator:	Bernard Feder, HGC Engineering
Proceedings Copyediting:	Cécile Le Cocq, Université du Québec (ÉTS)
Student Prices and Subsidies:	Hugues Nélisse, IRSST
Conference Website:	Jérémie Voix, Université du Québec (ÉTS)

CONFERENCE WEBSITE: **awc.caa-aca.ca**

Bienvenue à Winnipeg!

Semaine canadienne de l'acoustique

8 au 10 octobre 2014

Musée canadien pour les droits de la personne
Photo de Josel Catindoy, courtoisie de Tourism Winnipeg

Comité organisateur

Présidente de la conférence :	Karen Turner, Protec Hearing Inc.
Directeur scientifique :	Ramani Ramakrihnan, Ryerson University
Coordinateur exposants :	Bernard Feder, HGC Engineering
Correction des actes :	Cécile Le Cocq, Université du Québec (ÉTS)
Bourses et prix édudiants :	Hugues Nélisse, IRSST
Site web de la conférence :	Jérémie Voix, Université du Québec (ÉTS)

SITE DE LA CONFERENCE : awc.caa-aca.ca

Sponsors / Commanditaires

Exhibitors / Exposants

Éditorial invité
Guest Editor's note

Bienvenue à Winnipeg !

Le comité organisateur de la conférence se réjouit de vous accueillir à Winnipeg pour la Conférence annuelle 2014 de l'Association canadienne d'acoustique, du 7 au 10 octobre. Nous espérons qu'avec cet emplacement central, il sera facile pour les participants de partout au Canada de participer. Winnipeg est une ville d'arbres et de rivières, et les couleurs d'automne peuvent y être spectaculaires. Le Fairmont Winnipeg sera notre centre, situé en plein cœur de Winnipeg, entouré de restaurants et de théâtres, et à seulement quelques pâtés de maisons de "The Forks", un quartier d'histore et de loisirs, situé à la jonction des rivières Rouge et Assiniboine. «The Forks» est aussi maintenant l'emplacement du nouveau Musée canadien des droits de l'homme, qui sera le site de la réception du mardi soir. La réception mettra en vedette une grande surprise amusante qui convient particulièrement à notre conférence et au thème de l'acoustique. Il y aura aussi une tournée disponible le mercredi soir au Price Research Center North, qui est un lieu de test et de développement acoustique de pointe, l'un des plus avancé en Amérique du Nord. Je sais que beaucoup d'entre vous voudront prendre cela en compte.

Nous avons d'excellentes présentations prévues pour vous sur une grande variété de sujets

Welcome to Winnipeg!

OThe Conference Organizing Committee looks forward to welcoming you to Winnipeg for the 2014 Canadian Acoustical Association annual conference, October 7-10. We hope this central location will make it easy for participants from across Canada to attend. Winnipeg is a city of trees and rivers, and the fall colours can be spectacular. The well-appointed Fairmont Winnipeg hotel will be our venue, right in the heart of Winnipeg, surrounded by restaurants and theatres, and just a few blocks away from "The Forks" historical and recreation area, at the junction of the Red and Assiniboine rivers. "The Forks" is also now the home of the amazing new Canadian Museum of Human Rights, which will be the site for our Tuesday evening reception. The reception will feature an entertaining surprise that is uniquely appropriate to the venue and our Acoustics theme. There will also be an available tour on Wednesday evening of the Price Research Center North, which is a cutting edge acoustical testing and development facility, one of the most advanced in North America. I know that many of you will want to take this in.

We have some excellent presentations planned for you on a wide variety of acoustical topics, and featuring special keynote speakers Claude Alain, acoustiques, et mettant en vedette des conférenciers d'honneur, tels Claude Alain, Lola Cuddy, et Per Hiselius, avec de nombreux autres intervenants distingués. Aussi bien l'assemblée générale annuelle que le banquet auront lieu le jeudi soir à l'hôtel Fairmont. Pour ceux d'entre vous qui voudraient participer à d'autres évènements en dehors de l'horaire de la conférence, vous pouvez choisir entre Sherlock Holmes au Manitoba Theatre Centre; Dinosaurs au MTS Centre; Victoria Crosses of Valour Road au Manitoba Museum of Man and Nature; et Dali au Winnipeg Art Gallery - le tout à seulement quelques enjambées de l'hôtel Fairmont Winnipeg! Tout proche se trouve également l'étonnante nouvelle exposition d'ours polaires au Assiniboine Park Zoo.

Nos sincères remerciements à tous les membres du comité organisateur pour leur dur labeur, et en particulier à Ramani Ramankrishnan, notre directeur scientifique, pour l'organisation du programme, à Bernard Feder pour l'organisation de l'exposition et des commandites, à Cécile Le Cocq pour la relecture et la révision des actes de la conférence, à Hugues Nélisse pour les prix et bourses des étudiants, à Jérémie Voix pour la gestion du site de la conférence, ainsi qu'à Frank Russo et Christian Giguère pour leurs innombrables conseils.

Rendez-vous à Winnipeg!

Karen Turner
Présidente de la conférence

Lola Cuddy, and Per Hiselius, along with many other distinguished presenters. Both the Annual General Meeting and the Banquet will take place on Thursday night at the Fairmont. For those of you wanting to take in some events outside of the conference schedule, you can choose from Sherlock Holmes at the Manitoba Theatre Centre; Dinosaurs at the MTS Centre; Victoria Crosses of Valour Road at the Manitoba Museum of Man and Nature; and Dali at the Winnipeg Art Gallery – all just a few blocks from the Fairmont Winnipeg! A short ride away is the amazing new polar bear exhibit at Assiniboine Park Zoo.

Our special thanks to all the members of the Organizing Committee for their hard work, in particular to Ramani Ramankrishnan, our technical chair for the scientific program, to Bernard Feder for the management of the exhibit and sponsorship, to Cécile Le Cocq for the copyediting of the conference proceedings, to Hugues Nélisse for student awards and subsidies, to Jérémie Voix for the conference website management, as well as to Frank Russo and Christian Giguère for their continuous support.

See you in The 'Peg!

Karen Turner
Conference Chair

THE FAIRMONT WINNIPEG

Rest easy knowing your conference has been organized at The Fairmont Winnipeg. Over 20,000 square feet of meeting and banquet facilities accommodate up to 1,100, providing the most diverse and flexible meeting space of any downtown Winnipeg hotel. Guests enjoy our warm ambiance, exceptional service, and dining in the Velvet Glove—the only CAA/AAA Four Diamond rated restaurant between Toronto and Calgary—or unwinding in the Lounge at the Fairmont. You'll appreciate the hotel's immediacy to Winnipeg's vibrant arts and nightlife districts. Step outside and you are within minutes' walk of the Centennial Concert Hall, the historic Exchange District, The Forks, CanWest Global Park and the MTS Centre. The hotel's ideal location is further enhanced by its link to an indoor concourse of shops.

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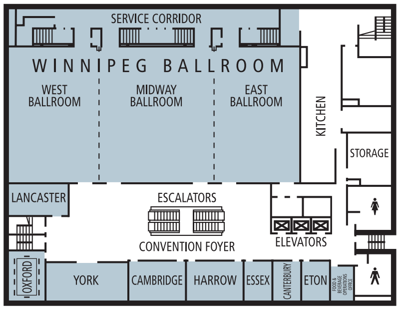
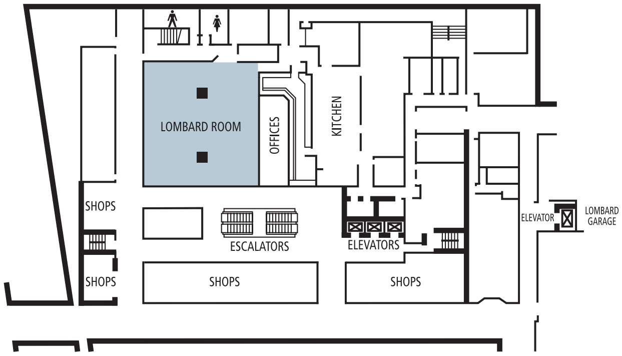
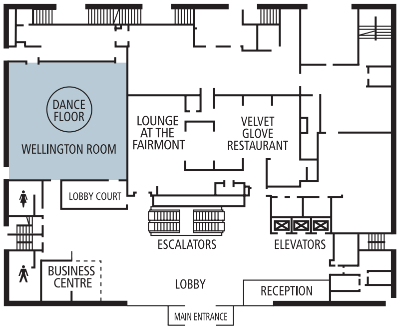
Mezzanine Level



Concourse Level



Lobby Level



ROOMS	SIZE	SQUARE FEET	HEIGHT	THEATER STYLE N.HT	THEATER STYLE W.HT	CLASS ROOM N.HT.	CLASS ROOM W.HT.	BOARD ROOM	ROUND TABLE N.HT.	ROUND TABLE W.HT.	DANCE & HEAD TABLE	RECEPTION	DANCE FLOOR
Winnipeg Ballroom West	65′ x 143′	9295	14	1100	1000	–	–	–	850	800	600	1000	–
East	65′ x 44′	2860	14	350	300	200	150	–	240	180	150	250	20x30
Midway	65′ x 55′	3575	14	450	400	300	225	–	300	250	170	350	35x40
West	65′ x 44′	2860	14	350	300	200	150	–	240	180	150	250	20x30
Wellington Ballroom**	55′ x 55.5′	3052.5	12	350	300	180	140	–	250	190	140	250	400sq.ft
Lombard Room*	58′ x 52′	3016	10	200	150	–	–	–	200	170	150	200	YES
HARROW, ESSEX & CANTERBURY	18′ x 54′	972	10	100	90	60	50	50	70	60	–	80	YES
ESSEX & CANTERBURY	18′ x 27′	486	10	50	40	30	25	30	30	–	–	40	–
Harrow	18′ x 27′	486	10	50	40	30	25	30	30	–	–	50	–
Essex	18′ x 14′	243	10	20	15	16	12	15	20	–	–	20	–
Canterbury	18′ x 14′	243	10	20	15	16	12	15	20	–	–	20	–
York	18′ x 40′	720	10	80	60	36	32	40	60	40	–	80	–
Cambridge	18′ x 27′	486	10	50	40	30	25	30	30	–	–	50	–
Oxford	17′ x 24′	408	10	–	–	–	–	10	–	–	–	–	–
Lancaster	14′ x 28′	392	10	30	25	16	10	15	20	–	–	20	–
Eton	18′ x 14′	243	10	20	15	20	10	10	10	–	–	15	–

* Concourse Level

** Lobby Level

AVERAGE TEMPERATURES (Celsius/Fahrenheit)	°C/°F High	°C/°F Low
WINTER JANUARY–MARCH	-5/23	-15/5
SPRING APRIL–JUNE	17/63	0/32
SUMMER JULY–SEPTEMBER	30/86	17/63
FALL OCTOBER–DECEMBER	7/45	-11/12

ROOMS

- Total Rooms 340
- Twin 6
- Double/Double 111
- Queen 55
- King 145
- Suites 20
- Non-Smoking 273
- Special Access 5

FACILITIES

- Airport – Winnipeg International Airport – (6 miles/10 km)
- VIA Rail Station – Winnipeg – (0.6 miles/1 km)
- Business Center
- 1 lounge and 1 restaurant – capacity 75 to 150
- Indoor rooftop pool and health club
- In-room refreshment centers
- Parking – outdoor and covered valet

ACTIVITIES

- See Activities Chart for more detailed information

LOCAL ATTRACTIONS

- The Exchange District
- Forks Market & National Historic Site
- Royal Winnipeg Ballet
- Assiniboia Downs Racetrack
- Winnipeg Symphony Orchestra
- Fort Whyte Centre
- Manitoba Theater Centre
- Lower Fort Garry National Historic Site
- The Festival du Voyageur
- Folklorama Festival
- Manitoba Museum of Nature
- Planetarium
- Centennial Concert Hall
- CanWest Global Park
- IMAX Theater
- MTS Centre

CONFERENCE SCHEDULE

DAY ONE	WEDNESDAY 8 OCT	
8:50-9:OO	**Welcome**	
9:00-10:00	**Keynote Talk - My Ears Are Alight - Per Hiselius**	
10:00- 10:20	COFFEE BREAK	
	Room A	**Room B**
	HEALTHCARE FACILITIES - John Swallow	HEARING CONSERVATION - Christian Giguere
10:20-10:40	**J. C. Swallow and M. J. Wesolowsky:** Floor Vibration Considerations for Sensitive Equipment in Hospital, Medical, Pharma and Laboratory Facilities	**A. Behar, F. Russo and R. Segu:** Comfort from Hearing Protectors
10:40-11:00	**K. Van Wyk and K. Murphy:** FGI Guidelines for healthcare acoustic design	**A. Behar:** A New CSA Hearing Protection Standard
11:00-11:20	**J. C. Swallow and M. J. Wesolowsky:** Acoustic Design Considerations in Modern Hospital Design	**J. Voix and C. Le Cocq:** Experimental Study On Custom Earplug Retention In Real-World Situations
11:20-11:40	**K. Van Wyk and K. Murphy:** Hospital noise methodology and survey results	**C. Giguere and T. Kelsall:** Relationship Between Direct And Indirect Methods Of Estimating Sound Exposure When Communication Headsets And Hearing Protectors Are Worn
11:40-12:00	**L. Scannell, M. Hodgson, D. Ng and J. Rauscher:** Evaluation of Objective and Subjective Acoustical Quality in Office-Type Health-Care Facilities	**J. Voix:** Did you say "Bionic" Ear?
12:00-1:15	LUNCH	
	ENGINEERING ACOUSTICS/ NOISE CONTROL - Peter van Delden	SPEECH SCIENCES - John Easling
1:15-1:35	**R. Ramakrishnan and N. Shinbin:** Parametric Analysis of Elbow Silencers	**J. Esling, C. Coey and S. Moisik:** Auditory Categories & Laryngoscopic/Ultrasound images in the "iPA Phonetics" App
1:35-1:55	**E. Mouratidis:** End Reflection Loss in Real Duct Systems	**B. Gick:** Some Concepts Underlying An Embodied Theory Of Speech Sounds
1:55-2:15	**I. Sabourin and B. Zeitler:** Comparison of Airborne and Impact Sound Insulation Improvement due to Adding Toppings on Homogeneous Floors	**M. Schellenberg and J. McDonough:** Using Discrete Cosine Transformations to Characterize Tones in Two Athabaskan Languages
2:15-2:35	**R. Ramakrishnan and V. Seharwat:** Sound Propagation from Wind Farms	**R. Greiss, J. Rocha and E. Matida:** Validation Of A Finite Element Continuum Model Of Vocal Fold Vibration
2:35-2:55	**N. Moeller:** Exploring the Impacts of Consistency in Sound Masking	
2:55-3:15	COFFEE BREAK	
	AERO-ACOUSTICS - Joanna da Rocha	PHYSICAL ACOUSTICS/ULTRASOUND - Bill Gastmeier
3:15-3:35	**R. Ramakrishnan, R. Dumoulin and P. Waudby-Smith:** Acoustic Simulation of Large Turning Vanes	**A. Elhelaly, M. Hassan and A. Mohany:** Localization of Acoustic Emission Source in Plates Using Wavelet Transform
3:35-3:55	**J. van Blitterswyk, and J. Rocha:** Prediction and Measurement of Turbulent Boundary Layer Wall-Pressure Fluctuations on the Surface of a Single Panel at Low Mach Numbers	**S. Iakovlev, C. Furey D. Pyke and A. Lefieux:** Transient Radiation By A System Of Two Cylindrical Shells
3:55-4:15	**S. Ghinet, A. Price, M. D. Alexander, A. Grewal and V. Wickramasinghe:** Closed Door Flight Test Investigation Of Cabin Noise Exposure In The NRC Bell 412 Helicopter	**D. P. Singh, A. Upmanyu:** Acoustical Investigations of Molecular Interactions in Polymer Solution of PAN/Clay Nano composites and DMSO
4:15-4:35	**S. Ghinet, A. Price, M. D. Alexander, A. Grewal and V. Wickramasinghe:** Open Door Flight Test Investigation Of Cabin Noise Exposure In The NRC Bell 412 Helicopter	**M. Shaaban, and A. Mohany:** Control of Acoustic Resonance in Shallow Rectangular Cavities Using Surface Mounted Blocks
4:35-4:55	**A. Mohany and R. Ramakrishnan:** Generation of High-intensity High-Frequency Noise	
5:00-5:30	**CAA STANDARDS MEETING (ROOM TBA)**	

DAY TWO	**THURSDAY 9 OCT**	
9:00-10:00	**Keynote Talk - Use it or Lose it: The benefit of musical training on the aging auditory brain - Claude Alain**	
10:00- 10:20	COFFEE BREAK	
	HEARING ACCESSIBILITY AND AGING - Kathy Pichora-Fuller	UNDERWATER ACOUSTICS - Garry Heard
10:20-10:40	**N. Newall and V. Menec:** Age-Friendly Communities Principles and Initiatives	**G. Heard, R. Dittman, N. Pelavas and G. Schattschneider:** Underwater Ice Cracking Noise Measurements In Yellowknife Bay, Great Slave Lake, Near The Detah Ice Road
10:40-11:00	**K. Pichora- Fuller:** Auditory and Cognitive Aging: Implications for Hearing Accessibility	**P. Van Delden, E. Stolp, A. Farag and S. Ghose:** Crickets: Temperature Dependent Background Sound
11:00-11:20	**H. Goy, K. Pichora-Fuller and P. van Lieshout:** Aging Voices And Speech Intelligibility: Implications For Communication By Older Talkers	**C. D. S. Tollefsen, D. D. Ellis and S. Pecknold**: Model-Data Comparisons For Transmission Loss And Reverberation During TREX13
11:20-11:40	**B. Chapnik and W. Gastmeier:** Recent Trends In The Acoustics Design Of Institutional Facilities	**J. lakziz, S. Ouaskit, O. Sofiane and R. Elguerjouma:** Finite Difference Time Domain Method for Acoustic Waves in Attenuate and Absorptive Medium for Layered Underwater Acoustic Environments
11:40-12:00	**D. Paccioretti:** Beyond Hearing Aids; Technologies To Improve Hearing Accessibility For Older Adults With Hearing Loss.	**T. J. LeBlanc and J. Fawcett:** Perceptual Feature Extraction For Target Classification On A Biosonar Dataset
12:00-1:15	LUNCH	
	SPEECH SCIENCES - Wladyslaw Cichocki	ARCHITECTURAL ACOUSTICS - David Quirt
1:15-1:35	**M. Noguchi:** Perception Of Intervocalic Consonant Clusters By Japanese Listeners	**D. Quirt:** A New Approach to Building Acoustics Regulation in Canada
1:35-1:55	**J. Turner, N Netelenbos, N. Rosen and F. Li:** Stop Consonant Production By French-English Bilingual Children In Southern Alberta	**I. Sabourin, B. Zeitler and D. Quirt:** Supporting Better Noise Control in Canadian Buildings
1:55-2:15	**N, N. Ellaham, C. Giguere and W. Gueaieb:** A New Research Environment For Speech Testing Using Hearing-Device Processing Algorithms	**R. Ramakrishnan, J.J.S. Kim, J.A. Smith and R. Roos:** Effect of Compression on Acoustic Performance of Fibrous Materials
2:15-2:35	**M. Kiefte:** Detection Of Deviations In Straight Formant Transitions	**R. C. da Silva and S. Paul:** Evaluation Of Different Materials As Protective Floor Covering In-Situ Impact Noise Measurements
2:35-2:55	**O. Valentin, M. Sasha and J. F. Laville:** Use Of Auditory Steady-State Responses In Measuring The Attenuation Of Hearing Protection Devices	**R. Ramakrishnan and T. Yousefi:** Pressure Drop Evaluation of Silencers
2:55-3:15	COFFEE BREAK	
	SPEECH SCIENCES- Wladyslaw Cichocki	HEARING SCIENCES - Frank Russo
3:15-3:35	**W. Cichocki, S. Selouani and Y. Perreault:** Measuring Rhythm In Dialects Of New Brunswick French: Is There A Role For Intensity?	**G. Nespoli, P. Ammirante and F. Russo:** Musicians Show Enhanced Neural Synchronization at Multiple Timescales
3:35-3:55	**T. Mitsuya, K. G. Munhall and D. W. Purcell:** Compensatory Vowel Formant Production With Artificial Voicing Source In Response To Real-Time Auditory Perturbation	**J. S. Arsenault, Y. He, G. M. Bidelman and C. Alain:** The Impact of Context on Auditory Stream Segregation Using Speech Sounds
3:55-4:15		**W. Wong, A. Namasivayam and P. van Lieshout:** Impact Of Auditory Attention On The Efferent Auditory System In The Absence Of Real Auditory Targets
3:00-4:30	**CSA STANDARDS MEETING (ROOM TBA)**	
4:30-5:30	**ANNUAL GENERAL MEETING**	
6:30-8:30	**BANQUET**	

DAY THREE	**FRIDAY 10 OCT**	
9:20-10:20	**Keynote Talk - Brain systems for music: exploring musical memory in aging and dementia - Lola Cuddy**	
10:20-10:40	COFFEE BREAK	
	ENGINEERING ACOUSTICS/ NOISE CONTROL - Serguei Iakovlev	ENGINEERING ACOUSTICS/ NOISE CONTROL - Ivan Sabourin
10:40-11:00	**S. Iakovlev, J. Rubio and P. Diez:** Modeling Acoustic Response Of Structurally Complex Shell Systems	**B. Murta, C. Hoeller, I. Sabourin and B. Zeitler:** Measurement of structural reverberation times for calculation of ASTC in upcoming NBCC
11:00-11:20	**A. Omer and A. Mohany:** The Effect Of High Frequency Vortex Generator On The Acoustic Resonance Excitation In Shallow Rectangular Cavities	**B. Logawa and M. Hodgson:** Innovative Ways to Make Cross Laminated Timber Panels Sound-Absorptive
11:20-11:40	**T. Kelsall:** Low Sound Levels Measured in Remote Areas	**O. Sadek, M. Shaaban and A. Mohany:** Suppression of Acoustic Resonance in Piping System Using Passive Control Devices
11:40-12:00	**G. Atkinson and G. Redman:** Determining Train Velocity from Sound Level Measurements	**F. Babic, A. Rodrigues and M. Salim:** Rooftop Noise Impact Investigation to the Community
12:00-1:00	**LUNCH & STUDENT PRESENTATION AWARDS**	

My Ears Are Alight

Per Hiselius, Ph.D. - 3M, Sweden

The capability of the human hearing is phenomenal. The dynamic range it can handle, the frequency range it covers, and not the least its ability to detect and identify speech and other signals in the presence of interfering sounds is astonishing. In our daily life we use this capability in many ways. We use it for speech communication as well as for alerts and alarms. We use it for analysis of devices and machines, e.g. our computers and cars. Is it on ? Does it sound normal, or is something wrong ? We also use it for various forms of entertainment. However, there's a flip side to the great capability of the human hearing. From an engineering point of view it poses challenges when designing buildings, machines, and devices such as phones, computers, mp3-players, headphones, microphones, et cetera. And although we have a phenomenal ability to understand speech also in challenging situations, we often mishear or misunderstand - we cannot trust our hearing. The human hearing is also quite easily damaged. I will present old and new results related to the capability of our hearing, and some of the challenges related to the same.

Bibliographic Notice :

Per Hiselius is an acoustics specialist within 3M. He holds a M.sc. in Engineering Physics and a Ph.D. in Acoustics from Lund University, Sweden. He started his professional career in 1992 working with the acoustics of hearing protectors. In 2005 he joined Sony Ericsson Mobile Communications to work with audio solutions within the area of mobile telephony. In the beginning of 2013 Per joined 3M, and is again working within the field of hearing conservation, with a special interest in speech communication.

Keynote speaker Per Hiselius (right) introduced by Jérémie Voix (left)

Use it or Lose it : The benefit of musical training on the aging auditory brain

Claude Alain, Rotman Research Institute, Baycrest Centre - Department of Psychology, University of Toronto

As we grow older, we often experience difficulty understanding what a person is saying in the presence of other sounds (e.g., television, music, other people talking). Such age-related declines in listening are a major challenge for hearing science and medicine because of their wide prevalence. Furthermore, hearing aid technologies have so far been unable to effectively alleviate this problem. Here, I will present studies that have investigated the role of musical training as a mean to mitigate age-related decline in difficulties understanding speech in noise. Behavioral and neuroimaging studies provide converging evidence that musicians exhibit exceptional auditory skills that allow them to cope with age-related hearing loss better than non-musicians. In particular, continuous engagement in musical activities throughout adulthood is associated with slower age-related decline in understanding speech in noise. Neuroscience research has shown that musical training enhances central auditory processing, which can compensate for peripheral hearing loss. The benefit of musical training on the aging auditory brain is exciting and it opens new avenues for developing new remediation programs and improving current rehabilitation protocols aimed at helping older adults in noisy environments.

Bibliographic Notice :

Claude Alain graduated from Université du Québec à Montréal in 1991. He then moved to the VA Medical Center at Martinez, California for his postdoctoral training where he worked with Dr. David Woods until 1996. Dr. Alain is currently a senior scientist at the Rotman Research Institute at Baycrest Center for Geriatric Care and Professor in Psychology at the University of Toronto. He received several awards including Canadian Institutes of Health Research Scholarship and the Premier Research Excellence Award from the Government of Ontario. His research is in the field of cognitive neuroscience and focuses on the brain processes supporting auditory scene analysis. Dr. Alain is using a combination of neuroimaging techniques (e.g., EEG, MEG and fMRI) to investigate which, and how, different brain areas work together during auditory scene analysis.

Keynote speaker Claude Alain (right) introduced by Kathleen Pichora Fuller (left)

Brain systems for music : exploring musical memory in aging and dementia

Dr. Lola L. Cuddy, Queen's University

Music and acoustics share a long interdisciplinary history. Methods from psychoacoustics can be usefully employed to measure sensitivity to musical patterns and to assess memory for music. The presentation will describe studies of the extent to which people with brain injury or neurodegenerative disease can process music. We have explored patterns of relatively preserved and impaired abilities in stroke and dementia. In specifying these patterns, we will learn much about how patients with dementia can appreciate music in everyday and therapeutic situations, as well as learn about the relationship between particular cognitive functions and healthy and impaired neural systems.

Biographical Notice

Lola Cuddy is professor emerita of Psychology at Queen's University, Kingston. She completed her doctoral thesis in musical pitch perception at the University of Toronto. At Queen's she founded the Music Cognition laboratory, of which she is Director, and developed an international reputation for her publications on psychology of music. Her most recent research involves a project demonstrating preserved musical memory in Alzheimer Disease. Dr. Cuddy is the editor of Music Perception, a journal published by University of California Press. Dr. Cuddy's research has been funded by the Natural Sciences and Engineering Council of Canada, the GRAMMY Foundation®, and the Alzheimer Society of Canada (Dr. Albert Spatz Award).

Keynote speaker Lola L. Cuddy (right) introduced by Frank A. Russo (left)

Editor's Note

AWC Proceedings: what you don't want to know

Many things must be considered to achieve perfectly referenced conference proceedings in today's digital world. You can trust your Editorial Board to handle it all perfectly and spare you the humdrum details, but in case you were wondering about the process behind the screen (pun intended), here are some things you do need to know.

The editorial board will produce two versions of the September issue of the Canadian Acoustics journal (JCAA) in order to include the proceedings of the Acoustics Week in Canada (AWC) conference.

- A pre-conference (PRECONF) electronic issue will be sent by email to all authors/presenters, registered attendees, sponsors and exhibitors of the AWC conference.
- A post-conference (POSTCONF) hardcopy will be mailed to all JCAA journal subscribers and journal advertisers.
- After the conference, the proceedings will be moved from the AWC's database to the JCAA's database for regular online access and proper indexing using your favorite digital scholar tools.

In order for authors to have their two-page article published in the PRECONF electronic issue, they must:

1. submit their abstract before the ABSTRACT deadline and their abstract must be accepted by the AWC Technical Chair;
2a. submit their 2-page summary article before the ARTICLE deadline and their article must be accepted by the AWC Technical Chair;
3a. have their 2-page article accepted by the JCAA copyeditor before the PRECONF deadline;
4a. register at least one of the authors for the AWC conference before the PRECONF deadline. In the case where the submitting author is not registered for the conference, the email of the registered author should be typed in exactly the way it was entered by the registered author, for proper internal cross-referencing.

Failure to comply with steps 2a., 3a. or 4a. would result in not having the article, but only the abstract, available in the PRECONF electronic issue. The author(s) will be given a second chance to have the 2-page article published in the POSTCONF proceedings issue, if they:

3b. have their 2-page article accepted by the JCAA copyeditor before the POSTCONF deadline;
4b. register at least one of the authors for the AWC conference before the POSTCONF deadline. See comment above in 4a, if the submitting author is not the registered one.

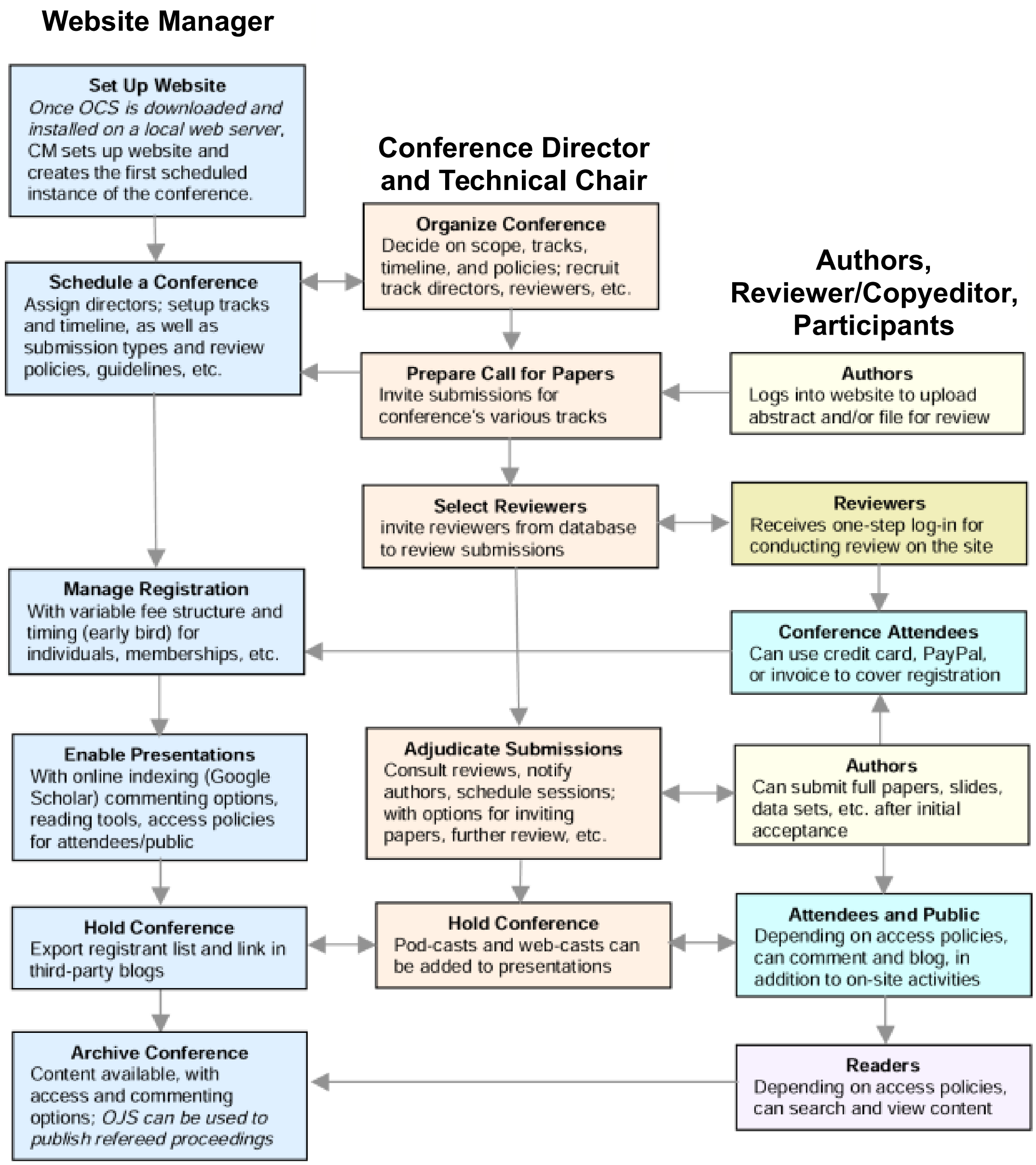
Finally, to have their two-page paper published in the POSTCONF proceedings issue as well as all subsequent publishing through the JCAA's online system, all authors must:

5. have at least one of the authors present the paper during the AWC conference.

For illustration purposes, here are the AWC 2014 planned and effective timeline (available on AWC website under "*Timeline*" at http://awc.caa-aca.ca/index.php/AWC/AWC14/schedConf/timeline)

ABSTRACT	June 15 (Postponed to June 30)
ARTICLE	August 1 (Postponed to August 31)
PRECONF	September 4 (Postponed to September 8)

And in case, you really want to learn more about how all this interact, here is some precious informations, adapted from the Open Conference System (http://ocs.sfu.ca/) that is now used by CAA.



Looking forward to meeting you all in Winnipeg!

Jérémie Voix
Editor

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AEROACOUSTICS - AÉROACOUSTIQUE

THE EFFECT OF HIGH FREQUENCY VORTEX GENERATOR ON THE ACOUSTIC RESONANCE EXCITATION IN SHALLOW RECTANGULAR CAVITIES

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1 Introduction

Flow over cavities can be a source of severe noise and/or excessive vibration when the oscillations generated by the shear layer formation in the cavity are coupled with the acoustic waves in the accommodating enclosure. This phenomenon is known as flow-excited acoustic resonance. Enormous research efforts have been made to mitigate the undesirable effects associated with this phenomenon [1]. The basic concept of attenuating the flow-excited acoustic resonance in cavities is by preventing/damping the formation of the shear layer over the cavity mouth, which results from the flow separation at the upstream edge, this can be achieved by placing a cylinder near to the upstream edge. The vortex shedding created by the cylinder interferes with the shear layer developed in the cavity, this interference can dampen the shear layer in the cavity and hence suppresses the acoustic resonance excitation. This interference is affected by many factors including the cylinder location and diameter, which are investigated in this paper.

2 Experimental and computational setup

The experimental setup consists of a test section made of 25.4 mm thick acrylic with a cavity installed at 330 mm from the test section inlet. The cavity dimensions are 127x127x127 mm which yields a length over depth ratio of 1.0. The inlet of the test section is attached to a bellmouth to help stabilize the flow and minimize the pressure drop. The outlet of the test section is attached to a diffuser that connects with a suction side of a centrifugal blower that is driven by a 75 horsepower motor controlled by a variable frequency drive and can achieve a maximum flow velocity of 155 m/s with a turbulence level less than 1% in the base case without any cylinder attached. A cylinder with a diameter of 4.57 mm which yields a ratio L/d of 27.78, where L is the cavity length and d is the cylinder diameter, was installed in different locations near the upstream edge of the cavity. Different locations in terms of vertical and horizontal distances were tested. In the results section the locations will be identified with the Cartesian coordinates assuming the tip of the sharp edge is the (0.0, 0.0) point, and the upstream direction is the negative x direction, while the downstream is the positive x direction. Another two cylinders with diameters of 3.81 and 6.35 mm, which yield L/d ratios of 33.33 and 20; respectively, were also investigated. Measurements of the acoustic pressure are taken from the cavity floor by means of a flush mounted microphone and analyzed using a Labview program with a sampling rate of 10 kHz and each signal is averaged 60 times which correspond to 60 seconds in real time.

The numerical simulation was performed using ANSYS Fluent 14, the simulation was divided into two parts, firstly, a steady state flow using K-epsilon turbulence model to obtain the average main parameters, secondly, unsteady state flow using Spallart Almaras Detached Eddy Simulation model, which is known in the literature to have a good prediction for flows over cavities. The mesh used consisted of approximately 169,000 quadrilateral cell for a Reynolds number of 43974.

3 Results and discussion

The effect of the cylinder on the resonance excitation is assessed by comparing each case when the cylinder attached at different locations with the base case which is the bare cavity without any cylinder attached. In the base case it is found that the first three shear layer modes are developed with Strouhal number of 0.46, 0.98, and 1.44; respectively. The values of the acoustic pressure are observed to be highly intensified when the resonance is materialized with values exceeding 2000 Pa and a lock-in region is observed.

Figure 1 shows the effect when the cylinder is located at a height of 8 mm from the test section bottom wall, the resonance excitation is significantly suppressed and the acoustic pressure is reduced to around 500 Pa. It is also observed that the resonance excitation is shifted to higher velocities, this shift is observed with most of the locations investigated. However, locating the cylinder at 50.8 mm upstream of the cavity results in shifting the resonance excitation to higher velocities without any suppression effect observed. Similar behaviour is observed when the cylinder is located at a height of 12.7 mm, where locations (-25.4, 12.7), (0.0, 12.7), and (25.4, 12.7) are able to suppress the resonance excitation and to keep the acoustic pressure at around 1000 Pa. Generally, it is observed that locating the cylinder at 25.4 mm from the cavity upstream edge has comparatively the best performance. In terms of the vertical distance, locating the cylinder closer to the bottom wall of the duct improves the suppression effect.

The cylinder diameter effect is investigated as well. It is found that increasing the cylinder diameter enhances the suppression mechanism, this is shown in Figure 2, where the cylinder is located at (-25.4, 12.7), it can be seen in the figure that the cylinder with a diameter of 6.35 mm can maintain the acoustic pressure below 150 Pa over the entire flow range investigated. To understand the interaction between the cylinder vortex shedding and the cavity shear layer, a 2D numerical simulation is carried out. For each cylinder location the velocity profiles obtained from the steady simulation at different distances along the cavity are

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compared. Also, the vorticity fields are compared to spot the development of the vortices at the cavity mouth.

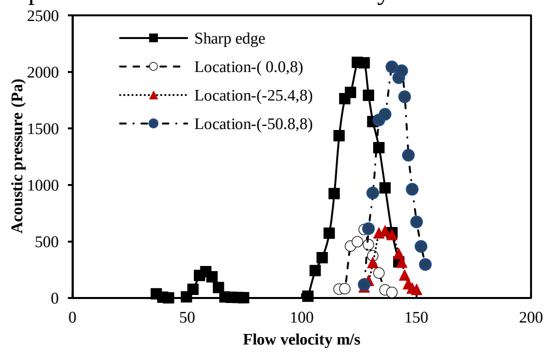

Figure 1 : comparison between base case and cylinder (d=4.57 mm) at locations of height 8 mm

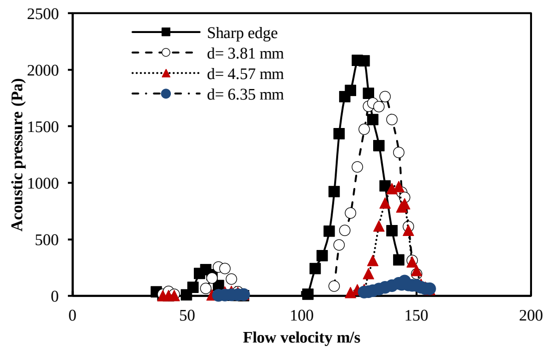

Figure 2: effect of cylinder diameter on suppressing the acoustic resonance excitation at location (-25.4, 12.7)

It is found that when the cylinder is located at a height of 25.4 mm, the wake of the cylinder does not interact with the cavity shear layer. It is noteworthy that this location experimentally was less effective in suppressing the resonance excitation. Moving the cylinder lower, to a height of 12.7 mm, at the same horizontal distance, 25.4 mm, is slightly changing the profile of the shear layer. The location at height of 8 mm which is found to be very effective experimentally is clearly influencing the shear layer at the cavity mouth, this can be seen in Figure 3. The wake of the cylinder interacts with the shear layer at the cavity mouth, introducing different profile for the shear layer. The cylinder wake can alter the shear layer in the cavity and increase the momentum thickness of the shear layer which result in more stability and less susceptibility to acoustic resonance excitation as discussed by Bruggeman et al.[2].

The effect of the horizontal location of the cylinder on the shear layer cannot be easily noticed from the shear layer thickness. However, the vorticity fields of each case obtained from the unsteady simulation show the progress of the vortex shedding and the interaction with the cavity shear layer. From the vorticity field it is observed that the cylinder vortex shedding interacts with the cavity shear layer and sends the main vortices developed by the shear layer into the cavity floor, hence, distracting the downstream impingement which results in interrupting the normal oscillations feedback cycle. This observation is evident for the locations that were found to be effective in suppressing the acoustic resonance experimentally, this can be seen in Figure 4 and Figure 5.

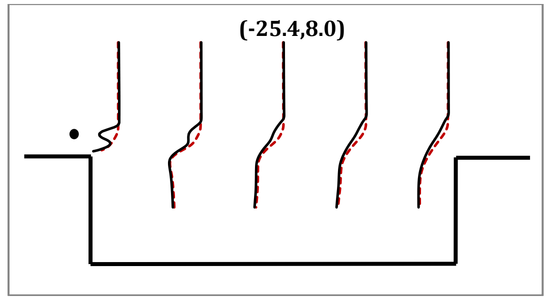

Figure 3: velocity profiles at x/L =0, 0. 2, 0.4, 0.6, and 0.8, base case (red dotted line), cylinder located at (-25.4, 8.0) (black solid line)

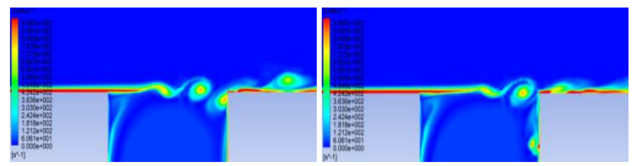

Figure 4:vorticity fields for base case at t= 0.08 & 0.1 sec

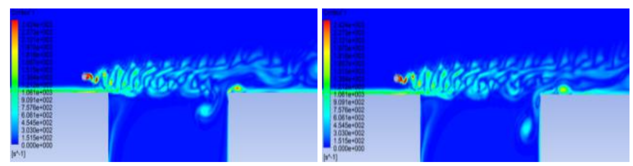

Figure 5: vorticity fields, cylinder location (-25.4,12.7) at t= 0.08 & 0.1 sec

4 Conclusion

Attaching a cylinder near the upstream edge of the cavity can be an effective technique in suppressing the acoustic resonance excitation. The location of the cylinder has a major influence in the effectiveness of this method, specifically, the vertical location of the cylinder from the bottom wall of the duct. Increasing the diameter of the cylinder generally improves the performance of the method, however, this should be tuned carefully to avoid exciting other modes by the cylinder wake itself. In most of the cases the acoustic resonance excitation is observed to be delayed to higher velocities, this shift is observed to be dependent on the location of the cylinder. The numerical simulation shows how the cylinder increases the momentum thickness of the shear layer at the cavity mouth, which results in more stable shear layer. From the vorticity fields, it is observed that the cylinder vortex shedding interacts with the cavity shear layer and distracts the normal oscillation cycle by altering the vortices impingement at the downstream edge.

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Prediction and Measurement of Turbulent Boundary Layer Wall-Pressure Fluctuations on the Surface of a Single Panel at Low Mach Numbers

Jared Van Blitterswyk* and Joana Rocha†
Department of Mechanical and Aerospace Engineering, Carleton University, Ottawa, Ontario

1 Introduction

The characterization and prediction of Turbulent Boundary Layer (TBL)-induced sound for aircraft applications has been investigated for several years [1]. It has been shown that a primary source of cabin noise during cruise conditions is induced by TBL wall-pressure fluctuations [2]. Early investigations provided insights into the characteristics of wall-pressure fluctuations, and several researchers developed semi-empirical models to predict the wall-pressure spectra at a single point. Such models have become integral components in advanced analytical frameworks for continuum models [3], and numerical approaches, such as Statistical Energy Analysis (SEA), as they provide the frequency distribution of the TBL excitation over the structure in question. Two semi-empirical, single-point frequency spectrum models, are reviewed and compared to experimental wall-pressure fluctuations measured on a rigid panel, in a wind tunnel facility, for Mach numbers (M) of 0.06, 0.09 and 0.12. The measured wall-pressure spectra are normalized by TBL variables to investigate spectral similarity over the range of Mach numbers.

2 Semi-Empirical Models

2.1 Efimtsov's Model [4]

Efimtsov developed two models using extensive flight testing and wind tunnel experiments, covering a range of subsonic and supersonic Mach numbers. His most recent model has the form of Eq. (1). In Eq.(1), f is the frequency, δ is the predicted TBL thickness, ρ is the freestream density, $U_\tau = U_\infty\sqrt{C_f/2}$ is the friction velocity, C_f is the friction coefficient, U_∞ is the freestream velocity, $Sh = 2\pi f\delta/U_\tau$ is the Strouhal number, $Re_\tau = \delta U_\tau/\nu_w$ is the Reynolds number based on wall shear stress, and ν_w is the kinematic viscosity at the wall. The empirical constant, α, has a value of 0.01, $\beta = [1 + (Re_{\tau o}/Re_\tau)]^{1/3}$, and $Re_{\tau o} = \delta U_\tau/\nu$.

$$\Phi(f) = (2\pi)^2\alpha U_\tau{}^3\rho^2\delta\frac{\beta}{(1+8\alpha^3Sh^2)^{\frac{1}{3}} + \alpha\beta Re_\tau\left(\frac{Sh}{Re_\tau}\right)^{\frac{10}{3}}} \quad (1)$$

2.2 Goody's Model [5]

More recent efforts by Goody were directed at modifying the Chase-Howe model to better agree with experimental measurements from a collection of sources. Goody's model has the form of Eq. (2).

In Eq. (2), τ_w is the wall shear stress, ν is the freestream kinematic viscosity, and $R_\tau = (U_\tau\delta/\nu)\sqrt{C_f/2}$ is the ratio of unsteady pressure time scales.

$$\Phi(f) = \frac{6\pi(2\pi f\tau_w)^2\left(\frac{\delta}{U_\infty}\right)^3}{\left(\left(\frac{2\pi f\delta}{U_\infty}\right)^{\frac{3}{4}} + 0.5\right)^{3.7} + \left(1.1R_\tau{}^{-0.57}\left(\frac{2\pi f\delta}{U_\infty}\right)\right)^7} \quad (2)$$

3 Method

A thick acrylic panel (thickness of 0.019 m) was simply-supported within the lower chamber of a custom, two-piece, noise reduction test section. The panel-chamber combination created a smooth lower surface for the test section with overall dimensions of 0.80 m x 0.48 m x 1.83 m. The upper chamber is lined with an acoustic foam to reduce the noise intensity from the wind tunnel motor and its control unit. Wall-pressure spectra were measured with a flush-mounted microphone, located 0.86 m from the start of the test section. A second, reference, microphone was placed at the same streamwise location, but spaced 50.8 mm in the spanwise direction, to temporally filter propagated noise from the wind tunnel fan and structure. The microphone array consisted of two, ¼ in.-diameter, Brüel and Kjaer 4944A type microphones. The microphones were outfitted with standard grid caps and custom caps with 0.5 mm pinhole diameters. Wall-pressure spectra measured with the grid cap configuration are truncated above 3 kHz, due to resonance. Only pinhole measurements were corrected using the Corcos correction [6] based on the condition that $\delta^+ = \delta U_\tau/\nu \leq 160$ [7].

4 Results and Discussion

The measured spectra are normalized using a mixed combination of inner and outer boundary layer parameters: τ_w, U_τ, and δ, to investigate spectral self-similarity for M = 0.06, 0.09 and 0.12. The normalized spectra, measured using the pinhole and grid cap microphone configurations, are shown in Fig. 1a and Fig. 1b, respectively. The non-dimensional spectra, measured with both configurations, collapse well over the mid-frequency range ($10 < 2\pi f\delta/U_\tau < 100$), but less so for low frequencies. The outer-layer scaling set ($\Phi(f)U_o/q_o^2\delta^*$ as a function of $2\pi f\delta^*/U_o$), were found to be equally acceptable over these ranges. A Mach number dependence is shown by the normalized spectra in Fig. 1a, for $2\pi f\delta/U_\tau \geq 10^2$, as the spectra did not collapse under any combination of inner, mixed or outer scaling variables. The spectra in Fig. 1b exhibit a much stronger self-similarity for $2\pi f\delta/U_\tau > 10$; however, this may be a result of scalable microphone

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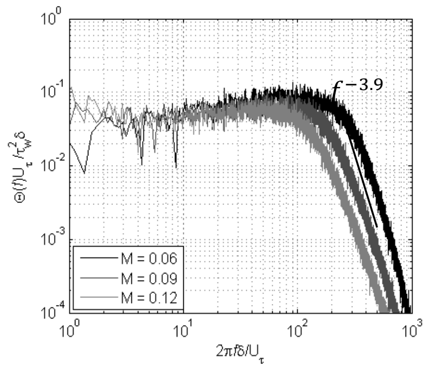
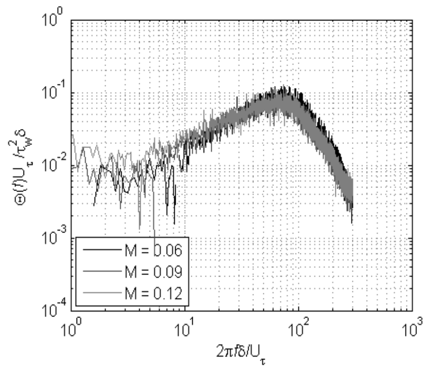
attenuation, disguised as flow-similarity. The steep spectral decay for $2\pi f\delta/U_\tau \geq 10^2$ shown by both data sets, is an indication of a not fully-developed boundary layer. This region, describing wall-pressure energy from the logarithmic region of the boundary layer, should exhibit a much shallower decay ($\propto f^{-1}$) but instead, the spectra exhibit a decay rate proportional to $f^{-3.9}$ (Fig. 1a). This decay rate is typical of the transition from overlap to high-frequency regimes.



a)



b)

Figure 1: Normalized single-point wall-pressure spectrum as measured with a) pinhole cap, and b) grid cap configurations

Figure 2 shows a comparison between measured and predicted spectra at M = 0.12. One can observe that low- and mid-frequency (f < 1 kHz) spectrum is best predicted by Efimtsov's model. Goody's model over-predicts spectral energy above 50 Hz, but most accurately captures the decay rate over higher frequencies (f > 1 kHz). No model accurately predicts the steep roll-off beginning around 1 kHz, and only Goody's model predicts a spectral peak. Both models account for an appreciable contribution from the overlap region (1 kHz < f < 50 kHz), which is not exhibited in the measured spectra; the absence of which, is believed to be the root cause for the discrepancies between experimental and predicted spectra above 1 kHz. Although Efimtsov's model less accurately predicts spectral decay rates above 1 kHz, it shows a much better agreement with experimental data over the low- and mid-frequency ranges and therefore, appears to be most appropriate for predicting wall-pressure fluctuations at low Mach numbers.

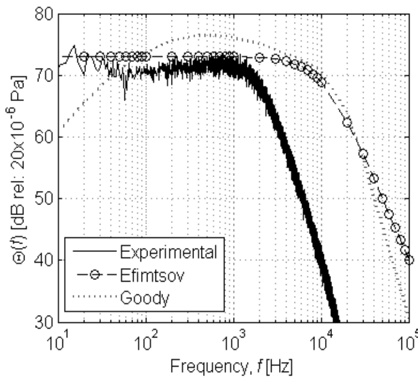


Figure 2: Wall-pressure spectrum for M = 0.12, as measured with the pinhole microphone, compared to existing semi-empirical models

Conclusions

The Efimtsov model is shown to best predict spectrum levels over the low- and mid-frequency ranges, while the overlap spectral energy decay rate is best predicted by the Goody model. For future work, the applicability of models by Goody and Efimtsov, and other models, will be re-evaluated with a fully developed TBL. A trip system is being installed in the wind tunnel to artificially develop the boundary layer, so that scaling variable dependencies can be better evaluated.

Acknowledgements

The authors are grateful for the support from the Natural Sciences and Engineering Research Council of Canada, and collaboration with Bombardier Aerospace.

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GENERATION OF HIGH-INTENSITY – HIGH-FREQUENCY NOISE

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2 - Department of Architectural Science, Ryerson University, Toronto, M5B 2K3

1 Introduction

Acoustic reverberation chambers are used to qualify space flight hardware to high sound pressure levels. High intensity sound generators that are capable of emitting sound power levels in excess 160 dB produce the levels mainly in the low frequency range and the high frequency levels are due to non-linear spill over. In space craft testing, high levels in high frequency testing may be required. Different mechanisms such as whistles, valves and nozzles are available to produce high frequency levels in the ranges of 140-150 dB. Resonant cavity scheme is one such method that produces strong sound levels. Heller et.al. studied flow induced oscillation in shallow cavities [1]. The main focus here is the generation of high intensity sound levels in contrast to suppression of noise levels, such as the work of Cattafesta et. al. [2]. A test program was conducted to evaluate the performance of cavity resonances for a range of operating parameters. The results of the measurement program will be presented in this paper.

2 Background

Typical spectra that are specified in space-vehicles testing is shown in Figure 1 below [3]. It can be seen that sound pressure levels between 140 – 150 dB are specified between 630 Hz and 8000 Hz frequency bands.

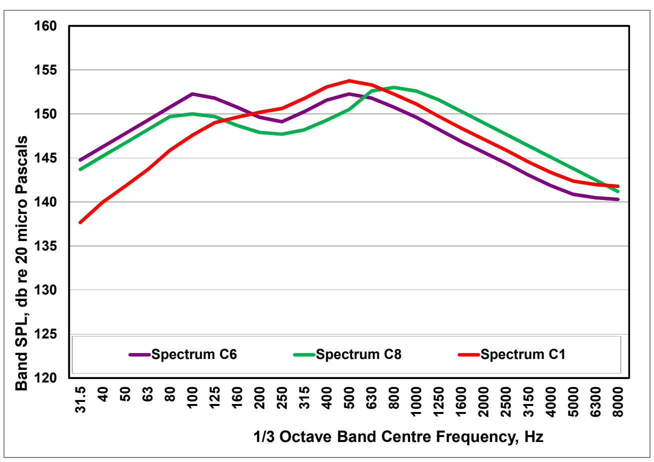

Figure 1: Required spectra for space vehicle testing [from Ref.3].

Heller et. al. used a shallow cavity in wind tunnel test section, as shown in Figure 2, to study resulting pressure and acoustic fluctuations [1]. The experiments showed the tested cavities were able to generate noise levels in the range of 120-140 dB in narrow frequency bands. The results of Reference 1 also showed that if one combined different edges and cavity designs, it may be possible to generate high intense sounds over several frequency bands.

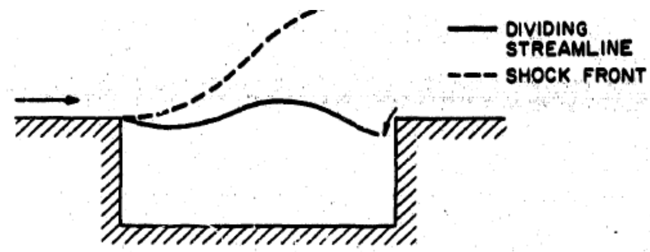

Figure 2: Schematics of flow over a shallow cavity [from 1].

3 The Experiment

The experimental setup, as shown in Figure 3, consists of 254 mm high and 127 mm wide test section made of 25.4 mm thick acrylic. A cavity with aspect ratio of 1.67, as illustrated in Figure 4, is attached at 330 mm from the inlet. The cavity dimensions are 127 x 127 x 63.5 mm. The acrylic test section is attached to a bell mouth that helps reducing the pressure drop at the inlet while maintaining a uniform flow inside the duct. The outlet of the test section is attached to a diffuser that connects with a suction side of a centrifugal blower. The blower is driven by a 75 horsepower motor controlled by a variable frequency drive and can achieve a maximum flow velocity of 155 m/s with a turbulence level less than 1%. The acoustic pressure is measured using a flush mounted microphone at the cavity floor, as shown in Figure 4.

Figure 3: Schematics of flow over a shallow cavity.

4 Results and Discussion

Figure 5 shows a contour plot of the pressure spectra for the acoustic resonance excitation in rectangular cavity with aspect ratio of L/D = 1.67; where L is the cavity length and

D is the cavity depth. The first and the second shear layer modes developed in the cavity excite the first acoustic cross-mode. The lines of the shear layer modes are illustrated in Figure 6(a), the slopes of these lines follow Strouhal number of each shear layer mode. The Strouhal number obtained from the experiments for the first shear layer mode is 0.48 and for the second shear layer mode is 0.94. Figure 6 is constructed from the contour plot of the pressure spectra shown in Figure 5. Each point in Figure 6 represents the amplitude and frequency of the vortex shedding component taken from the pressure spectra. In addition, it can be seen from Figures 5 and 6 that when the frequency of the shear layer mode comes close to the acoustic resonance mode, the process of flow-excited acoustic resonance is initiated and a lock-in region is observed.

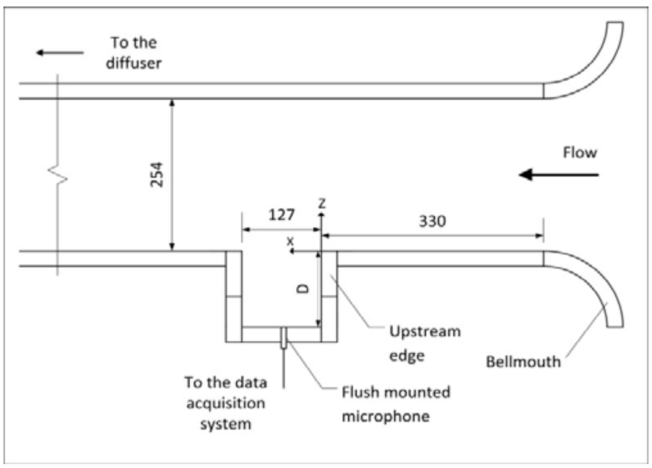

Figure 4: Schematics of flow over a shallow cavity.

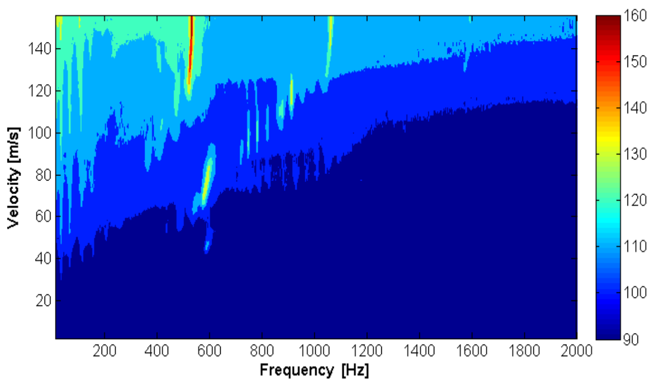

Figure 5: Contour plot of the pressure spectra measured at the cavity floor.

The frequency of the acoustic mode can be predicted from the formula,

$$f_a = nc / 2H \qquad (1)$$

where *c* is the speed of sound, *n* is the number of the acoustic mode excited and *H* is the total height of the test section. The first resonance occurs when the second shear layer mode excite the first acoustic mode at around 585 Hz and starts at flow velocity of 66 m/s. The second resonance occurs when the first shear layer mode excites the first acoustic mode at around 530 Hz and starts at flow velocity of 120 m/s. The excitation of the acoustic resonance by the first shear layer mode produces the highest acoustic pressure amplitudes of about 3300 Pa (i.e. 164 dB), as can be seen in Figure 6(b).

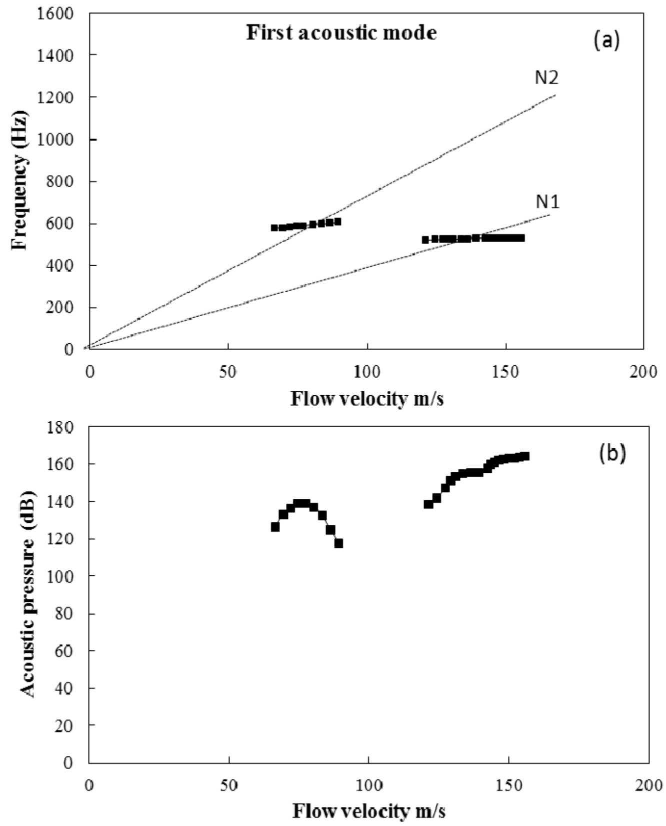

Figure 6: Experimental results: a) Cavity modes and b) Sound pressure levels, dB

5 Conclusion

Sound levels in excess of 140 dB in set frequency bands were generated by using shallow cavities. The simple experiment showed that high intense sounds can be generated in high frequencies. The current investigation could be modified by incorporating different cavity combinations to generate high sound levels in a wide range of high frequencies.

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CLOSED DOOR FLIGHT TEST INVESTIGATION OF CABIN NOISE EXPOSURE IN THE NRC BELL 412 HELICOPTER

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1 Introduction

Aircraft interior noise impairs crew communication and degrades comfort. The helicopter cabin acoustic environment consists of multi-tonal rotor noise, broadband noise, and high frequency transmission/hydraulic systems noise. Continuous exposure may lead to hearing loss if personnel hearing protection is insufficient or improperly worn. The evaluation of noise levels experienced by helicopter aircrew is essential for the selection of optimum hearing protection for enhancing aircrew safety and mitigating long term health issues. Aircrew cabin noise exposure was investigated by the National Research Council of Canada (NRC) on its Bell 412 research helicopter. Results were compiled to determine the acoustic performance of hearing protection devices. Sound Pressure Levels (SPL) were measured in the aircraft cockpit and aft cabin at three aircrew stations during a selection of standard flight maneuvers.

2 Flight test procedures and equipment

The objective of the flight test was to characterize the cabin noise exposure of aircrew during typical helicopter maneuvers. The NRC Bell 412, a civilian variant of the Royal Canadian Air Force's CH-146 Griffon tactical/utility helicopter, was instrumented for the investigation. The aircraft cabin retained its standard manufacturer acoustic treatments with testing performed with aft cabin doors closed. Noise measurements were recorded at three cockpit and aft cabin stations.

Standard procedures including ISO 5129 [1], ISO 9612 [2] and CSA Z107.56-06 [3] as well as MIL-STD-1294A [4] were applied in this investigation. These standards defined requirements for instrumentation, as well as test procedures for the measurement and reporting of sound pressure levels at crew and passenger locations under steady flight conditions. The flight test procedure adopted for this investigation was similar to that for the flight test performed with aft cabin doors opened. Details of this flight test plan and aircraft configuration (including microphone fixtures and instrumentation system) were presented in Ref [7].

2.1 Microphone mounting

In accordance to ISO 5129, the microphones were placed at fixed locations with custom designed mounts and a tripod to minimize measurement interference. The hands-free design eliminated the need for operators to grasp microphone extension rods during measurements. Windscreens were fitted over all microphones in order to prepare them for the open doors flight test described in Ref [7]. Note that the insertion loss of all windscreens was measured in advance in the absence of wind using a reverberant chamber.

2.2 Airworthiness considerations

During the flight test, aircrew wore the SPH 5CF flight helmet. For safety reasons, no sensors were attached directly to the aircrew. With respect to aircraft airworthiness requirements, the TTC MSSR-100C series miniature Data Acquisition (DAQ) System installed on the NRC Bell 412 helicopter was considered a non-essential item for normal flight operations. This DAQ system was configured as a self-powered, standalone unit operating on batteries. The unit was installed in the aircraft by strapping it to a passenger seat. Each functional module of the DAQ system was certified by TTC according to applicable MIL and non-government standards for aircraft flight test purposes. As integrated, the system was not part of the NRC Bell 412 critical flight instrumentation and did not interfere with aircraft operations.

3 Results

3.1 Flight data parameters

Flight testing was performed in Ottawa (Canada) over NRC test ranges and routes. Both acoustic and flight data were recorded for analysis. Flight test maneuvers consisted of ground interfacing, hovering, and steady forward flight conditions. Flight parameters including air data, inertial data, rotor speeds and torque data were recorded on the NRC Bell 412 research data acquisition system as follows:

- Barometric Altitude (**ALT)**, feet;
- Airspeeds (Indicated, **IAS**; True, **TAS**), knots;
- Attitudes (**Pitch**, **Roll**, **Heading**), degrees;
- Rotor speeds (Main, **MRRPM**; Tail, **TRRPM**), RPM;
- Main Rotor Mast Torque (**TQMast**), %

As presented in **Table 1**, the maximum altitude and true airspeed for the flight test occurred at 1800 feet and 135 knots, respectively. Pitch and roll attitudes did not exceed 6 degrees and 3 degrees, respectively. The baseline rotor-speed (324 RPM) varied approximately 1 %. Of significance in high frequency engine noise generation, the highest mast torque (83%) was recorded during high speed descent.

Table 1. Recorded Flight Data Parameters

Event	Alt (ft)	IAS (kts)	TAS (kts)	Pitch (Deg)	Roll (Deg)	Heading (Deg)	MRRPM (RPM)	TRRPM (RPM)	TQMast (%)
GROUND RUN	380.05	8.14	14.52	3.61	-0.21	278.40	323.71	1658.51	23.81
HOVER (H = 60)	434.09	10.05	15.72	6.15	-0.84	279.71	324.15	1660.78	72.60
CLIMB (V = 60)	1043.56	54.86	58.10	4.10	-2.38	37.60	325.04	1665.35	67.01
SLF (V = 100)	1390.68	91.96	96.67	0.04	-1.44	57.43	326.35	1672.06	59.70
SLF (V = 120)	1337.93	109.69	115.23	-0.80	-2.21	58.97	323.32	1656.52	78.22
DESCENT (V = 140)	1126.97	129.19	135.31	-3.10	-1.94	78.57	323.00	1654.90	80.64
CLIMB (V = 50)	1285.55	53.22	56.74	3.83	-2.18	241.35	324.95	1664.85	61.66
SLF (V = 90)	1782.17	92.21	97.65	0.29	-0.65	288.28	325.11	1665.69	63.16
SLF (V = 110)	1825.13	110.45	116.89	-0.81	-1.77	251.70	323.48	1657.31	78.23
DESCENT (V = 140)	1320.59	128.57	134.87	-3.43	-2.39	239.64	323.08	1655.28	83.29
HOVER (H = 35)	429.96	8.31	14.61	5.42	-2.20	281.87	324.04	1660.21	73.02
GROUND RUN	378.69	7.82	14.34	3.41	0.05	286.59	322.79	1653.83	24.40

3.2 Cabin noise exposure

The Sound Pressure Levels at three cabin crew stations were measured during flight testing. Pilot noise exposure was evaluated using Insertion Loss data, which was measured in the SPH 5CF flight helmet in accordance to procedures specified in ANSI/ASA S12.42-2010 [5] using the GRAS Acoustic Fixture 45CB.

The un-weighted and A-weighted Sound Pressure Level spectra at the pilot's helmet position for each flight condition are presented in **Figure 1**. The highest measured noise levels were due to the main rotor 8/rev tone. The maximum measured level of this main rotor harmonic (i.e. SPL of 108 dB) occurred in the 40 Hz 1/3-octave band and was recorded during a 120 knots level flight condition. Analysis of the A-weighted SPL of the acoustic data revealed a maximum level of 91 dBA in the 630 Hz 1/3-octave band during a descending flight condition.

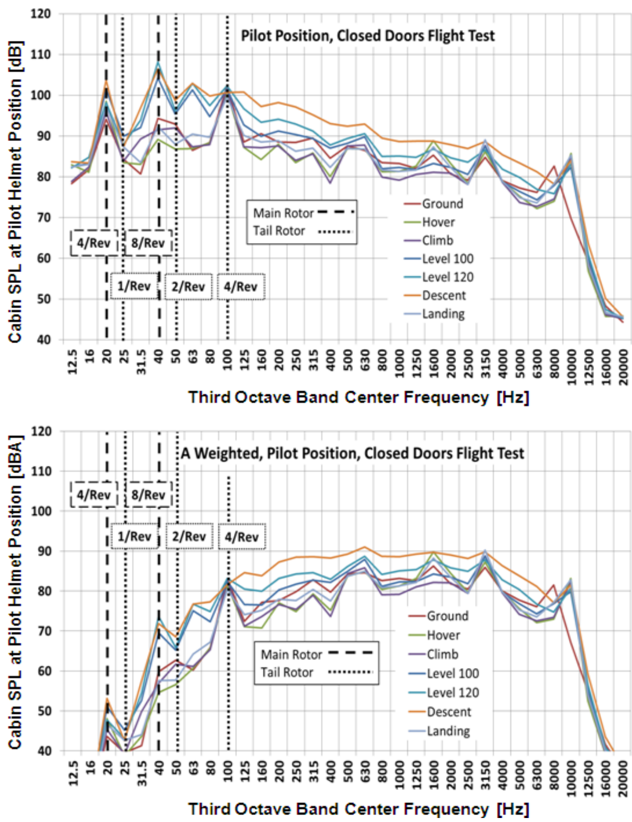

Figure 1. Cockpit Noise Spectra Near the Pilot's Helmet: (Top) Un-weighted SPL; (Bottom) A-weighted SPL

As depicted in **Figure 2** and determined previously in Ref [7], it was found that the SPH 5CF flight helmet provides acceptable noise attenuation of helicopter interior noise with the aircraft configured with its cabin doors closed. The maximum estimated sound pressure level exposure for a pilot wearing this helmet was found to be 71 dBA in the 100 Hz 1/3-octave band. This demonstrated that the flight helmet complied with Canada Labour Code, Part II. This standard dictated that the sound pressure level exposure in an aircraft for a period of 8 hours should be less than 87 dBA.

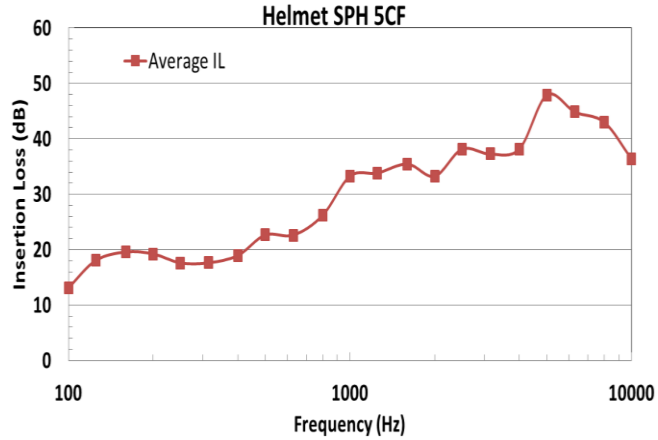

Figure 2. Insertion Loss of the SPH 5CF Flight Helmet.

4 Conclusions

The main objective of the work was to characterize the cabin noise exposure of aircrew during a range of flight maneuvers. Noise levels in the NRC Bell 412 helicopter configured with aft cabin doors closed were measured at three cockpit and aft cabin stations. Cabin noise was generated during flight maneuvers consisting of ground interfacing, hovering, and steady forward flight conditions.
At low frequencies, main and tail rotor tones were prevalent in many flight conditions. The 8/rev main rotor tone was prevalent with a maximum SPL of 108 dBA in the 40 Hz band recorded during high speed level flight. The maximum A-weighted SPL (91 dBA in the 630 Hz band) was recorded during descending flight. Pilot noise exposure was evaluated using Insertion Loss data measured in the SPH 5CF flight helmet using the GRAS Acoustic Fixture 45CB according to ANSI/ASA S12.42-2010 [5]. It was been determined that the SPH 5CF helmet provided acceptable attenuation for the measured cabin noise in compliance with the Canada Labour Code, Part II.

Acknowledgments

The authors are grateful of the valuable contributions made by NRC Aerospace helicopter test pilots Stephan Carignan and Timothy Leslie during this investigation.

References

[1] ISO 5129:2001, Acoustics - Measurement of sound pressure levels in the interior of aircraft during flight.

[2] ISO 9612:2009, Acoustics - Determination of occupational noise exposure - Engineering method.

[3] CSA Z107.56-06, Procedures for the measurement of occupational noise exposure.

[4] MIL-STD-1294A, Acoustical noise limits in helicopters

[5] ANSI/ASA S12.42-2010, Methods for the Measurement of Insertion Loss of Hearing Protection Devices in Continuous or Impulsive Noise Using Microphone-in-Real-Ear or Acoustic Test Fixture Procedures.

[6] Canada Labour Code, Department of Justice Canada

[7] S. Ghinet, A. Price, M. Alexander, A. Grewal, V. Wickramasinghe, Y. Chen, "Open door flight test investigation of cabin noise exposure in the NRC Bell 412 helicopter", Canadian Acoustics Week, 2014.

OPEN DOOR FLIGHT TEST INVESTIGATION OF CABIN NOISE EXPOSURE IN THE NRC BELL 412 HELICOPTER

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1 Introduction

Military helicopter missions involving rappelling, search and rescue, as well as the delivery of payloads and weapons require flight with cabin doors open. The cabin acoustic environment is comprised of noise that is mechanically and aerodynamically generated. In this aircraft configuration cabin noise levels may exceed the limits of aircrew hearing protection equipment. Aircrew cabin noise exposure was investigated in the National Research Council of Canada (NRC) Bell 412 research helicopter with its cabin doors open. The results were compiled to determine the acoustic performance of aircrew hearing protection equipment.

2 Flight test procedure

The objective of the flight test was to characterize the cabin noise exposure of aircrew during typical helicopter maneuvers with cabin doors open. The NRC Bell 412, a civilian variant of the Royal Canadian Air Force's CH-146 Griffon tactical/utility helicopter, was instrumented for the investigation. The aircraft cabin retained its standard manufacturer acoustic treatments. Noise measurements were recorded at three cockpit and aft cabin stations.

Requirements for test instrumentation, as well as procedures for the measurement and reporting of cabin interior sound pressure levels under steady flight conditions were obtained from ISO 5129 [1], ISO 9612 [2], and CSA Z107.56-06 [3] standards. In accordance to ISO 5129, the interior of the aircraft was unaltered with reference to the normal mission configuration. Seat backrests were set to their most upright position, while the number of occupants in the test aircraft was kept to the minimum (3) required to conduct the tests. To eliminate interference of sound propagation, there were no obstructions between the microphone locations and the aircrew positions. The positions of crew members included the pilot-in-command (right seat, cockpit), flight test engineer (left seat, cockpit), and the data acquisition operator (center seat, aft cabin).

Flight test sorties encompassed standard helicopter maneuvers categorized into three groups. Ground interface maneuvers included ground running, take-off, and landing on a paved tarmac. Stationary flight involved hovering in helicopter ground effect at an altitude of 50 feet. Steady airspeed maneuvers included climbing, level, and descending flight. The maximum flight speed and altitude attained during testing were 80 knots and 1800 feet, respectively. Flight conditions for each maneuver were maintained for a minimum of 60 seconds to provide steady acoustic environments suitable for recording. The duration of each sortie was approximately 45 minutes. The test aircraft and instrumentation suite are shown in **Figure 1**.

Figure 1. NRC Bell 412 (Top); Seating Position Microphones (Left); Acoustic Data Acquisition System (Right)

2.1 Flight test instrumentation

Instrumentation integrated into the aircraft included microphones and a recorder. Flight state parameters (such as air data, inertial data, rotor torque and speed) were recorded by the NRC Bell 412 research data acquisition system.

The Teletronics Technology Corp. (TTC MSSR-2010-SAR-2) portable battery-powered recording system was used for acoustic data acquisition. Three PCB Piezotronics microphones (ICP type PCB 378B02 with preamplifier type 426E01) were used for acoustic measurement. These microphones were calibrated according to standard test procedures using a GRAS Type 42AC sound calibrator.

The seated position microphone was located on the seat centreline. In accordance with MIL-STD-1294A, it was oriented with the vertical axis pointed upwards, located at a distance of 0.15m from the headrest, and 0.8 meters above the unoccupied seat cushion. The pilot microphone position was located at the seated pilot head height. With the pilot present and seated, it was located within 0.1 meters of the helmet position. The standing crew microphone position was located at 1.12 meters above the floor, which was equivalent to a seated crew forward leaning position. This microphone location replaced an unrealizable standing position (i.e. 1.65 meters above the floor) due to lack of clearance for a standing crew in the Bell 412 cabin.

2.2 Airworthiness considerations

During this flight test, aircrew wore the SPH 5CF flight helmet. For safety reasons, no sensors were attached directly to aircrew. The TTC MSSR-100C series Miniature Data Acquisition System was deemed a non-essential item for normal flight operations. Additional airworthiness considerations related to the data acquisition system were detailed in Ref [7].

3 Results

3.1 Cabin noise exposure

Sound Pressure Levels at the three cabin-crew stations were measured during a flight test sortie. Pilot noise exposure was evaluated using Insertion Loss data measured in the SPH 5CF flight helmet in accordance to procedures specified in ANSI/ASA S12.42-2010 [5] using the GRAS Acoustic Fixture 45CB.

Both un-weighted and A-weighted Sound Pressure Level (SPL) spectra at the pilot's helmet position in the cockpit for each flight condition are depicted in **Figure 2**. The low frequency region was dominated by the 4/rev harmonic of the main rotor tone. The maximum measured SPL of 120 dB occurred in the 20 Hz 1/3-octave band during both 80 knots level and descending flight conditions. Note that the most significant difference between the closed and open aft cabin door results occurred at the 4/rev frequency which was approximately 10 dB higher in the latter case. The measured A-weighted SPL data revealed a maximum of 92 dBA in the 3.15 kHz band during climbing flight conditions.

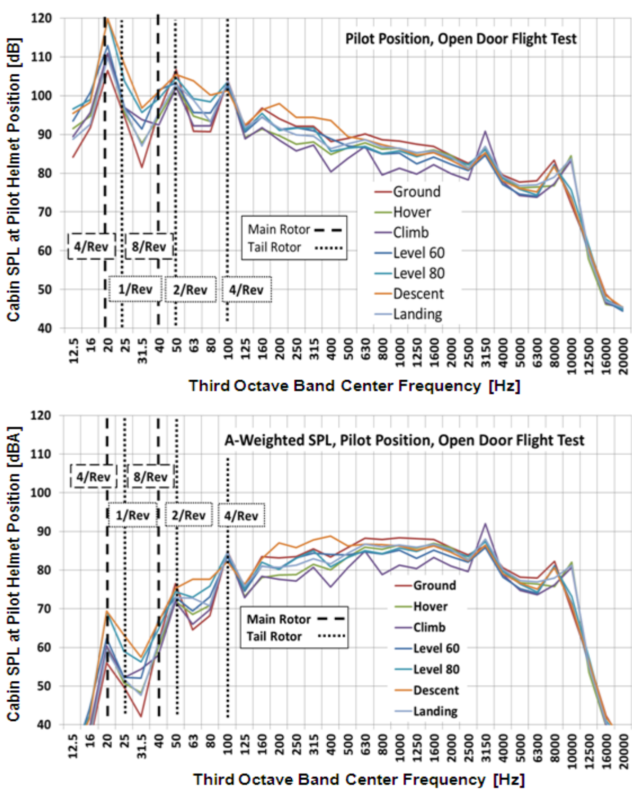

Figure 2. Cockpit Noise Spectra Near the Pilot's Helmet: (Top) Un-weighted SPL, (Bottom) A-weighted SPL

Using the Insertion Loss data for the SPH 5CF flight helmet which was collected during reverberant room testing (**Figure 3),** the maximum sound pressure level exposure for the pilot wearing this flight helmet was estimated to be 72 dBA. This indicates that the helmet provides acceptable attenuation of cabin noise with the helicopter configured with its cabin doors open, and demonstrates that the helmet complies with Canada Labour Code Part II which specifies that the sound level exposure in an aircraft over an 8 hour period should be less than 87 dBA.

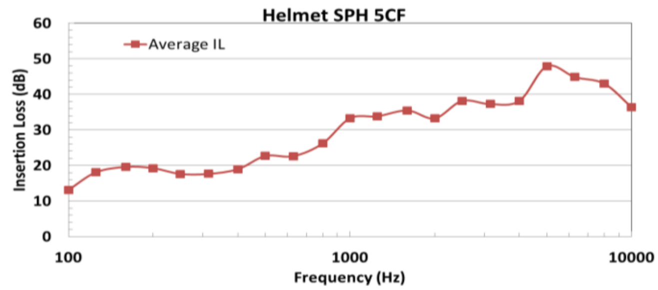

Figure 3. Insertion Loss of the SPH 5CF Flight Helmet.

4 Conclusions

In this investigation, the noise exposure of aircrew was characterized during typical maneuvers with helicopter cabin doors open. Noise levels in the NRC Bell 412 helicopter were measured at cockpit and aft cabin stations. At low frequencies, acoustics were dominated by the 4/rev main rotor tone for most flight conditions. The maximum SPL (120 dB in the 20 Hz 1/3-octave band) was measured during both level and descending flight. In the higher frequency range, the maximum A-weighted SPL (92 dBA at the 3150 Hz band) was measured during climbing flight. Pilot noise exposure for the SPH 5CF flight helmet, estimated using Insertion Loss data, demonstrated that this helmet provided acceptable attenuation cabin noise in compliance with the Canada Labour Code, Part II.

Acknowledgments

The authors are grateful to NRC Aerospace helicopter test pilots Stephan Carignan and Timothy Leslie for their valuable contributions during this investigation.

References

[1] ISO 5129:2001, Acoustics - Measurement of sound pressure levels in the interior of aircraft during flight.

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[7] S. Ghinet, A. Price, M. Alexander, A. Grewal, V. Wickramasinghe, Y. Chen, "Closed door flight test investigation of cabin noise exposure in the NRC Bell 412 helicopter," Canadian Acoustics Week, 2014.

Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Acoustic Simulation Of Large Turning Vanes

Ramani Ramakrishnan, Romain Dumoulin, Peter Waudby-Smith

Turning vanes are a must in large wind tunnels to provide the smooth transition of flow in the closed-loop circuit. The number of the large vanes can exceed ten. One of the methods to reduce fan noise levels is to treat the vanes with passive acoustic materials such as open-cell foam, and fibrous materials. The evaluation acoustic performance of the multi-unit vanes is complex and one has to resort to numerical methods such as FEM procedures. Two methods are used to evaluate the performance. The description of the methods and the results are presented in this paper.

ARCHITECTURAL ACOUSTICS - ACOUSTIQUE ARCHITECTURALE

A NEW APPROACH TO BUILDING ACOUSTICS REGULATION IN CANADA

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1 Introduction

A new approach to controlling sound transmission between adjoining units in residential buildings is proposed in the 2015 edition of the National Building Code of Canada (NBCC). The design objective is changing from a minimum STC for the wall or floor/ceiling assembly separating adjacent units to a minimum Apparent Sound Transmission Class (ASTC), which includes transmission of both direct and flanking sound. The design approach uses data from ASTM E90 laboratory measurements of direct transmission through wall or floor/ceiling assemblies together with flanking transmission data conforming to ISO 10848 as inputs to calculation procedures based on ISO 15712-1. This paper focuses on explaining the technical intent and form of the proposed new NBCC requirements.

2 The New Minimum Requirement

The minimum requirement changes from STC 50 for the separating assembly to ASTC 47 for sound transmission (including flanking transmission) between adjoining dwelling units. This limit was chosen to avoid significant increase in average cost of construction, while discouraging the use of construction details that seriously degrade system performance.

This should be recognized as a regulatory minimum, which many occupants would not consider to be satisfactory sound insulation. Many builders try to provide much better noise control. The supporting publications and calculation tools described in a companion paper [1] provide the resources to achieve this via systematic design.

3 Three Paths to show Compliance

There are three paths to establish compliance with the Code requirement. Rather than reproduce the requirements in Code language, the design approach is explained here in technical terms. For a more detailed explanation see NRC report RR-331 [2] which is also referenced in the NBCC.

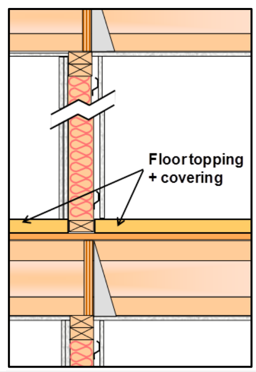
3.1 Show Compliance via Field Testing

A design is acceptable if its details replicate a system (separating assembly, flanking constructions, and junctions) that has been shown to provide ASTC 47 or better in field testing according to ASTM E336.

3.2 Show Compliance via Prescriptive Method

The section of the NBCC dealing with houses and small buildings provides a set of prescriptive details that are deemed to satisfy the ASTC requirement. For a limited set of separating constructions whose STC and fire resistance ratings are listed in tables in the NBCC, specific prescriptive requirements are provided for common generic flanking assemblies connected to the separating assembly at its edges.

An example for a generic wood-stud separating wall combined with wood-framed floor, ceiling, and side wall flanking assemblies is presented with simplified phrasing in Fig. 1 to indicate the nature of a typical set of prescriptive requirements.



- Separating wall with single row of 38 mm x 89 mm wood studs, with resilient metal channels on 1 side
- Separating wall has gypsum board, absorptive material in cavities, and spacing of studs and resilient metal channels as required for walls of types W5, W6, W10, or W12 in Table B.1 with STC 50 or greater
- Gypsum board on the wood-framed side walls and ceiling is interrupted or ends where it meets the framing of the separating wall
- Wood-I joists of connected floor not continuous across floor/wall junction, joists spaced 400 mm o. c. or more, with a subfloor of OSB or plywood at least 15.5 mm thick
- Floor topping and floor covering over the subfloor, with combined mass per unit area of 8 kg/m^2 or greater, on both sides of the junction
(e.g.- 16 mm hardwood strip flooring or an added layer of 16 mm OSB or plywood with any flooring on top)

Figure 1: Prescriptive details required to meet design objective (ASTC=47 or greater) between side-by-side spaces, for a specified combination of separating wall and attached flanking assemblies

The prescriptive requirements were based on calculations for sets of single path data tested according to ISO 10848, for typical combinations of closely-comparable connected assemblies. Construction details such as fastening gypsum board to the framing of flanking surfaces were assumed to be the worst-case variant consistent with approved practice, and a minimum improvement was identified for the most significant flanking path – providing a heavier floor surface in this case.

An appendix suggests some variants which could improve performance (such as choosing surfaces for the separating wall to increase Direct STC, or mounting the gypsum board ceiling on resilient metal channels) in order of their usefulness, but this prescriptive process gives no indication of the resulting ASTC.

3.3 Show Compliance via Design Method

The new Code requirements and ISO 15712-1 approach predicting the sound transmission from the same basic concept – combining the sound power transmitted directly through the separating assembly with the flanking transmission via first-order flanking paths at each edge of the separating assembly. To discuss this, it is useful to introduce the convention used in ISO 15712-1 for labelling the transmission paths, as illustrated in Figure 2.

Consider transmission from a source room at the left to the receiving room beside it. Each transmission path involves one surface in the source room (denoted by a capital letter) and one in the receive room (lower case). **D**irect transmission through the separating wall is path **Dd**. For each edge of the separating assembly there are three 1st-order flanking paths, each involving a surface in the source room and one in the receiving room, that connect at this edge: **Ff** from **f**lanking surface F to **f**lanking surface f, **Df** from **d**irect surface D to **f**lanking surface f, and **Fd** from **f**lanking surface F to **d**irect surface d in the receiving room.

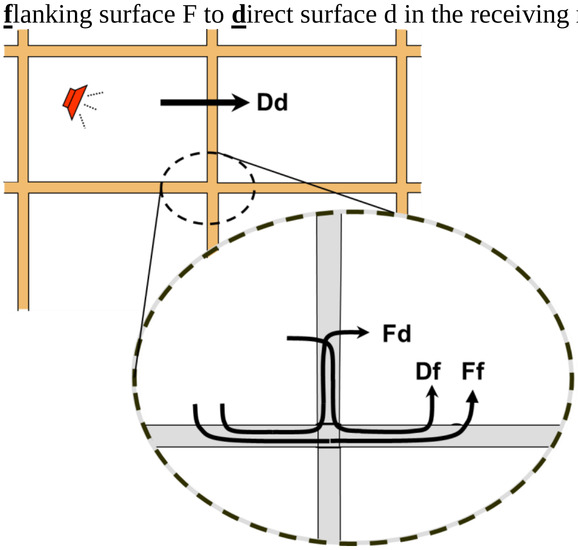


Figure 2: Labelling convention used in ISO 15712-1 for direct and flanking transmission paths

Note that "**F**" and "**f**" denote **f**lanking surfaces, whereas "**D**" and "**d**" denote the surface for **d**irect transmission, i.e. - the surface of the separating assembly. Each of these labels may apply to either wall or floor/ceiling assemblies, depending on orientation of the room pair.

In Canada, building elements (walls, etc.) are normally tested according to ASTM E90 and the Code requirements are given in terms of STC or ASTC ratings determined from the 1/3-octave test data, following the procedure in ASTM E413. Merging this ASTM context familiar to the building industry and to regulators with the ISO procedures now being added to the Code, requires new terminology, so "direct transmission loss" and "flanking transmission loss" have been introduced to provide consistency with ASTM terminology, but match the function of the direct and flanking sound reduction index, as defined in ISO 15712-1.

Section 4.1 of ISO 15712-1 defines a process to calculate apparent sound transmission by combining the sound power transmitted via the direct path and the twelve first-order flanking paths (See Figure 2). Equation 14 of ISO 15712-1 is recast here with different grouping of the paths, assuming rectangular room geometry and neglecting paths due to leaks, ducts, crawlspaces, etc., which should be controlled by normal good practice. The Apparent Sound Transmission Loss (ATL) between two rooms is the decibel expression of the sum of sound power due to Direct Sound Transmission Loss ($\mathrm{TL_{Dd}}$) through the separating wall or floor element and the sound power due to Flanking Sound Transmission Loss contributions ($\mathrm{TL_{Ff}}$, $\mathrm{TL_{Fd}}$, and $\mathrm{TL_{Df}}$) of the three flanking paths for every junction at the edges of the separating element:

$$ATL = -10\lg\left(10^{-0.1\cdot TL_{Dd}} + \sum_{edge=1}^{4}\left(10^{-0.1\cdot TL_{Ff}} + 10^{-0.1\cdot TL_{Fd}} + 10^{-0.1\cdot TL_{Df}}\right)\right)$$

This summation of transmitted sound power is valid for all building systems, but the remaining challenge is to find the right expressions to calculate the path transmission for the chosen building system and situation. The design procedure proposed for the NBCC constrains these choices, depending on the type of wall and floor constructions combined to form a complete building, as follows:

- ***For heavy homogenous types of construction*** such as concrete floors or concrete block walls, the NBCC design procedure determines the flanking sound transmission loss by either the Detailed or Simplified calculation procedures of ISO 15712-1. For input data, these calculations use sound transmission loss data (for the base wall and floor assemblies and for linings) measured according to ASTM E90.
- ***For lightweight steel- or wood-framed assemblies***, the NBCC design procedure substitutes experimental flanking data (treating flanking sound reduction index determined using ISO 10848 as Flanking Transmission Loss) for values calculated with ISO 15712-1. Either a detailed calculation using 1/3-octave-band data or a simplified procedure using the corresponding single-number ratings is permitted.

In either case, the calculation combines the sound power due to direct and flanking transmission in the same way.

Conclusion

Re-focusing the noise control requirements of Canada's building codes on the performance of the complete system should both avoid the worst designs and shift industry focus to optimizing the transmission paths that limit the ASTC.

References

[1] I. Sabourin, B. Zeitler and J.D. Quirt, Supporting Better Noise Control in Canadian Buildings, *AWC 2014*

[2] D. Quirt, B. Zeitler, S. Schoenwald, I. Sabourin, T. Nightingale, "RR-331: Guide to Calculating Airborne Sound Transmission in Buildings" *NRC Canada*, 2013.

SUPPORTING BETTER NOISE CONTROL IN CANADIAN BUILDINGS

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1 Introduction

A simplistic requirement for minimum STC of the wall or floor/ceiling assembly separating adjacent units in multi-family residential buildings has been used in North American building codes for over 50 years. Effective noise control requires replacing the traditional design objective with a requirement including the effect of flanking transmission, such as the Apparent Sound Transmission Class (ASTC). This paper discusses key projects at the National Research Council of Canada both to establish the technical infrastructure supporting such a change in Canada's building codes and to provide the tools needed by designers seeking to provide enhanced noise control.

2 Impediments to Change

Such a transition requires a supporting set of technical standards for measuring direct and flanking sound transmission for typical assemblies and junctions, plus a credible procedure for calculating system performance from these inputs.

Although using ASTM E336 to measure ASTC in a building is routine for Canadian consultants, predicting the ASTC due to the set of transmission paths in a building is much more complex, and there are no ASTM standards for measuring transmission of flanking sound (in lab or field) or for calculating ASTC.

However, ISO has developed a standardized framework for calculating overall sound transmission. ISO 15712-1 (aka EN-12354-1) has been used for over 20 years to support performance-based European code systems. It uses inputs from laboratory tests to characterize sub-assemblies (ISO 10848 and ISO's counter-parts of ASTM E90). But there are significant impediments to applying ISO 15712-1 procedures in a North American context:

1) ISO standards for building acoustics differ appreciably from the ASTM standards used by the construction industry in North America – both in their terminology and in the specific technical requirements for both measurements and ratings.
2) ISO 15712-1 provides reliable estimates for heavy homogeneous constructions such as concrete floors and masonry walls, but not for the lightweight steel- or wood-framed construction widely used for low-rise and mid-rise buildings in North America.

3 Supporting a Transition

A brief description of the proposed NBCC calculation procedures in a companion paper[1] indicates how the new building code straddles the ISO/ASTM divide, but glosses over some details of the calculation process. More importantly, it ignores the issue of finding the necessary laboratory test data to use as inputs for the calculations, and the practical need for software tools to ease the calculation process for both the design and regulatory communities. These shortcomings are addressed in supporting materials published or planned by NRC.

3.1 Guideline to clarify calculation details

The proposed Code directs users to document RR-331, "Guide to Calculating Airborne Sound Transmission in Buildings" for additional advice. A first version of this guideline was published in 2013, and updated editions are expected at the end of each year as content is refined and additional sections are approved by the industry steering committee.

This Guide presents extended descriptions of the calculation process (both simplified and detailed methods) for specific types of construction, and includes numerous benchmark examples of the calculations. It is intended mainly as a reference document for acoustical experts.

3.2 Data and examples for specific constructions

A set of reports are being prepared to present the data needed for the calculations and to provide guidance on spreadsheet calculations following the Simplified Method. These reports and the related experimental studies are being developed in collaboration with the industry associations representing specific construction materials, and are intended to provide guidance to a broad industry and regulatory audience.

These reports are based on the results of a series of large experimental studies performed in the NRC flanking transmission facilities with strong support from industry partners. These studies have characterized a broad sample of the generic constructions most commonly used in North American buildings. The plan is to publish a set of 5 documents with some overlap in content:

- RR-333, Apparent Sound Insulation in Concrete Buildings (2015),
- RR-334, Apparent Sound Insulation in Concrete Block Buildings (2014),
- RR-335, Apparent Sound Insulation in Cross Laminated Timber Buildings (2014),
- RR-336, Apparent Sound Insulation in Wood-framed Buildings (2015),
- RR-337, Apparent Sound Insulation in Steel-framed Buildings (2015).

The first of these documents (RR-334) was published in the summer of 2014 and the remainder will follow before the end of 2015. All will be reviewed and updated periodically, and can be downloaded from the NRC website at: http://nparc.cisti-icist.nrc-cnrc.gc.ca/npsi/ctrl?lang=en

Each of these reports presents sound transmission data pertinent to design calculations for buildings that include the construction cited in the title, plus worked spreadsheet examples combining that construction with other constructions with which it is commonly paired. The data presentation combines convenient summary pages giving single-number ratings (needed for the Simplified Method) in the body of the report with tables in the appendices giving the corresponding 1/3-octave-band data (needed for the Detailed Method). The worked examples illustrate the Simplified Method using single-number ratings, as illustrated in Figure 1.

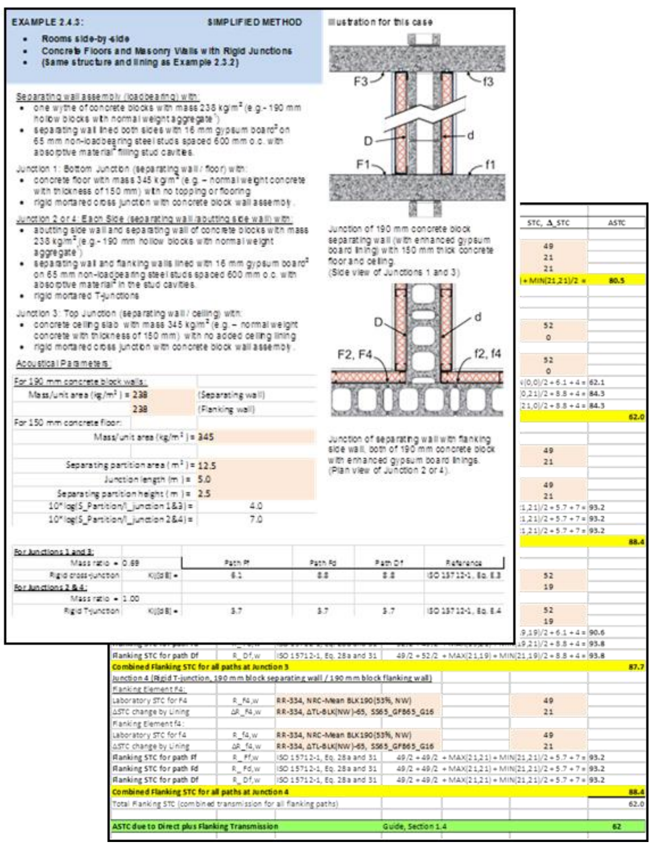

Figure 1: Worked example from RR-331 for a pair of side-by-side rooms. The 2-page example begins with drawings and text description of the assemblies and their junctions, followed by spreadsheet images showing all steps of the calculation

3.3 Software for calculation and data access

An online software tool called ***soundPATHS*** has been developed, and is currently being extended. The application ***soundPATHS*** provides a user friendly interface with drop-down menus to select the construction details, combined with a secure database that provides flanking transmission data for each junction and direct transmission data for the separating assembly. An image of the user interface (after elements have been selected for the displayed wall/floor assemblies and their junction) is shown in Figure 2.

Drop-down menus permit selection of the common generic building materials and sub-systems comprising each wall or floor/ceiling assembly. The calculation provides ASTC for the complete system together with direct STC for the separating assembly and Flanking STC for each junction at its edges. This identifies the weakest paths, and helps the designer to explore options to improve designs for higher ASTC performance and/or cost optimization.

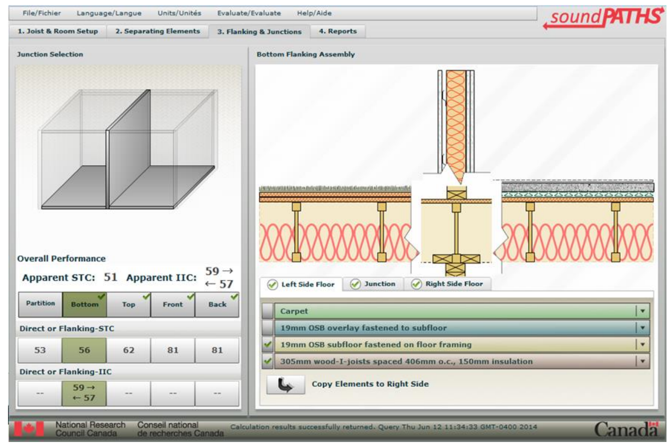

Figure 2: Image of the user interface for the ***soundPATHS*** online calculation of ASTC for a pair of side-by-side rooms with wood-framed wall and floor assemblies.

The current version of the ***soundPATHS*** software deals only with common Canadian wood-framed constructions. It can be accessed at: http://www.nrc-cnrc.gc.ca/eng/solutions/advisory/soundpaths/index.html

A ***soundPATHS*** upgrade that deals with other common types of wall and floor assemblies is under development. This will include a much-expanded supporting database using the data from the reports discussed in section 3.2.

Conclusion

Re-focusing the noise control requirements of Canada's building codes on the performance of the complete system should both avoid the worst designs and shift industry focus to optimizing the transmission paths that limit the ASTC. NRC will continue to pursue projects with industry partners to obtain and provide more generic data and to improve supporting calculation tools, to make this transition more manageable.

References

[1] J.D. Quirt and B. Zeitler, A new approach to building acoustics regulation in Canada, *AWC 2014*

[2] D. Quirt, B. Zeitler, S. Schoenwald, I. Sabourin, T. Nightingale, "RR-331: Guide to Calculating Airborne Sound Transmission in Buildings, *NRC Canada*, 2013.

[3] B. Zeitler, D. Quirt, S. Schoenwald, I. Sabourin, F. King, RR-334: Apparent Sound Insulation in Concrete Block Buildings *NRC Canada*, 2014.

PRESSURE DROP EVALUATION OF SILENCERS

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1 Introduction

Porous materials are usually applied in HVAC system ducts as passive silencers. The introduction of silencer materials such as baffles in the duct stream produces pressure drop to the fan system. The standard for pressure drop testing in silencers is ASTM E-477 [1]. The pressure drop measured in pascals becomes significant in elbow silencers where 90 degree turns are the norm. The pressure drop of elbow silencer systems were evaluated using a CFD method as well as simple ASHRAE procedures. The comparison of the two methods is presented in this paper.

2 ASHRAE pressure drop calculation

The pressure loss in the ducts is attributed to two types of pressure losses: the pressure loss due to the duct length and local pressure loss [1, 2]. They are calculated from the following equations.

$$\Delta PL = \sum_{i=1}^{n} f_i \rho \frac{L_i}{D_{hi}} \frac{V_i^2}{2} \quad (1)$$

$$\Delta PC = \sum_{j=1}^{m} C_j \rho \frac{V_j^2}{2} \quad (2)$$

where ΔPL and ΔPC are pressure drop for friction and local loss respectively, f_i is the friction factor and is a function of Reynolds number and roughness of the duct, ρ is the density of air in kg/m^3, L_i and D_{hi} are the length and hydraulic diameter of the duct in each section respectively, V_i and V_j are the flow velocity along the different sections for length pressure loss and in each section for local pressure loss, and C_j is the coefficient of dynamic loss. The friction factors, hydraulic diameters and Reynolds numbers can be evaluated from standard relationships.

3 Results and discussions

3.1 Test elbow silencers

Two elbow silencers were used to evaluate the pressure drop. The elbow silencer, schematically shown in Figure 1 has the following details.

Silencer	L1, in	L2, in	L3, in	R1, in	R2, in
Sample 1	135.4	270.8	15	3.42	20.57
Sample 2	135.4	270.8	15	8.7	15.27

Table 1: Silencer details

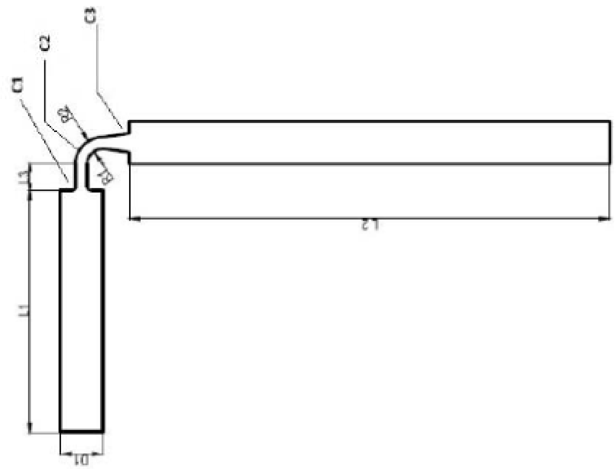


Figure 1: Details of a typical elbow silencer.

3.2 Pressure drop by ASHRAE

The dynamic loss coefficient can be obtained ASHRAE manuals and are shown in Table 2 for the two silencers. ε is the roughness of the duct an for the current experiments is equal to 1.5×10^{-4} m.

Silencer	C1	C2	C3	V(fpm)
Sample 1	1.14	1.3	0.32	500
Sample 2	5.33	1.52	9.3	1500

Table 2: ASHRAE coefficients of dynamic loss of the silencers.

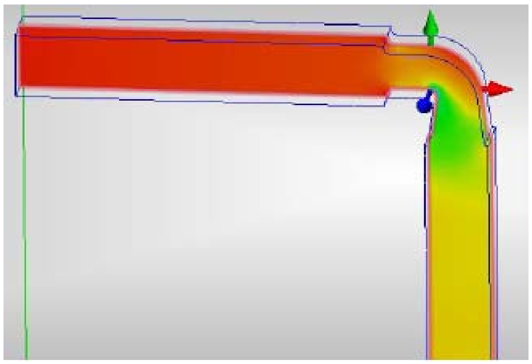
Calculated pressure loss based on the ASHRAE standard are presented in Table 3 below. It can be seen that the pressure losses due to the friction are very small compared to the local losses and can be neglected. Among the local pressure losses, the pressure losses in the elbows are greater than the contraction and expansion parts.

Silencer	ΔPC1	ΔPC2	ΔPC3	ΔPL1	ΔPL	ΔPL
Sample 1	4.4	9.9	1.24	0.18	0.36	0.02
Sample 2	186	713	325	1.4	2.8	0.15

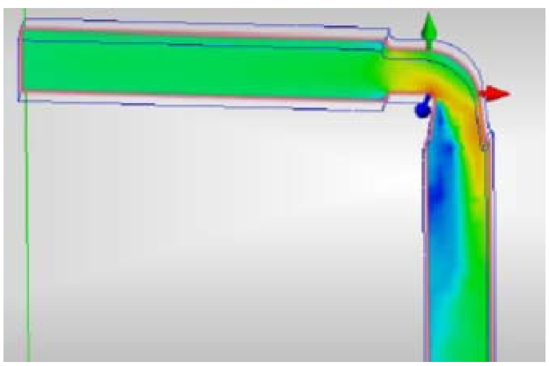
Table 3: Pressure losses, Pa, within the two elbow silencers.

3.3 Pressure drop calculation by numerical methods

Conventional CFD (computational fluid dynamics) software was used to calculate the pressure distribution and velocity distribution with the elbow silencers. Based on the pressure distribution, local pressure drop values were evaluated at locations along the entire centre-line length of silencer. The total pressure losses for sample 1 and sample 2 are 8.68 and 400 Pa respectively. By adding up the pressure losses in Table 3 it can be seen that the total pressure loss from ASHRAE standard for each case is much more than the numerical result. In order to understand this discrepancy the pressure variation along the centerline of the elbow silencer for sample 1 is depicted in Fig. 3.



a) Pressure distribution



(b) Velocity Distribution

Figure 2: Results for Sample 1

4 Discussion

The results of Figure 3 show that frictional losses vary smoothly while, local losses that have sharp reduction in pressure. It can also be seen that there is a pressure recovery after the minimum pressure.

Figure 3 shows the detail of the pressure reduction. Due to the pressure recovery, the pressure losses in numerical analysis, which is the pressure difference between point A and B, are not in agreement with total pressure losses from ASHRAE, which is shown as C1, C2, and C3 in this figure.

It's worth noting that the total of local pressure losses based on ASHRAE standard is,

$$\mathbf{\Delta PC_{total} = \Delta PC1 + \Delta PC2 + \Delta PC3 = 4.4+9.9+1.24 = 15.54\ Pa} \quad (3)$$

and the total local pressure loss based on the numerical analysis and by refer to Figure 3 is,

$$\mathbf{PA - PB = 10 - (-2) = 12\ Pa} \quad (4)$$

which is close to the ASHRAE result.

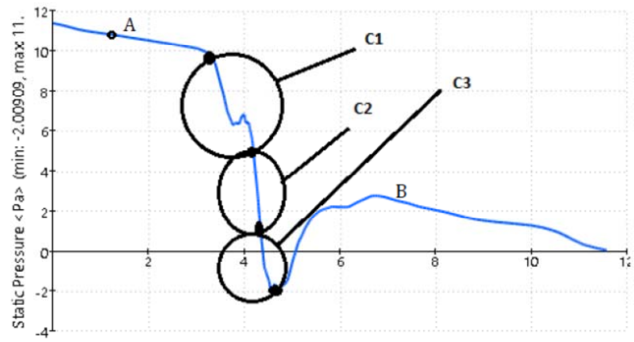


Figure 3: Pressure variation along the centre line of silencer sample 1.

5 Conclusion

ASHRAE standard can be used as an accurate reference for calculating the local pressure loss in various section of the duct. By adding up the local pressure losses, the total local pressure loss can be derived. The results are in agreement with total local pressure losses based on the numerical analysis. However, since there is a pressure recovery in the real case or/and in the numerical analysis, ASHRAE standard's results for total pressure loss are not in agreement with numerical analysis and cannot be used to calculate the total pressure loss. It means that ASHRAE standard can be used for calculating the pressure loss in each component separately and this standard is not able to consider the recovery effect.

Acknowledgments

We would like to acknowledge that the work was conducted under a research grant from the Ontario Centre of Excellence with Kinetics Noise Control as the industry partner.

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EVALUATION OF DIFFERENT MATERIALS AS PROTECTIVE FLOOR COVERING IN-SITU IMPACT NOISE MEASUREMENTS

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1 Introduction

Brazil's new building standard NBR15575 :2013 [1] - establishes requirements, criteria and evaluation methods to determine the performance of residential buildings in order to provide thermal, lighting, and acoustic comfort. For acoustic comfort, the standard specifies minimum performance values for airborne sound insulation and impact noise through by means of the parameters $D_{n,T,w}$ and $L_{n,T,w}$, respectively. Thus, in-situ measurements of the airborne sound transmission and impact noise are required to verify if the new buildings match the requirements established by the standard. According to ISO 140-7 :1998 [2] which defines methods for in- situ measurements of impact sound insulation of floors, it is necessary to use a standardized tapping machine (STM). As the hammers might damage the floor finishing a modified tapping machine (MTM), that uses a protective material between the floor and the hammers, is used as an alternative. The present study was carried out in a laboratory set-up in order to identify a suitable material to be used in the modified tapping machine to protect the floor from damage during the acoustic in-situ measurements.

2 Method

In order to find a suitable material to protect the flooring, tests were conducted in the impact sound transmission chamber of the Acoustic Laboratory at the Federal University of Santa Maria, in Brazil.

Following the recomendations of ISO 10140 :2010 [3], parts 1, 3 and 4 , ISO 717 :2013 [4], part and ISO 354 :2003 [5] (reverberation time of the receiving chamber) measurements of the impact sound pressure level were carried out exciting the different floors with a tapping machine modified by inserting different materials between the hammers and the floor finishings.

Three different materials were tested in a laboratory set-up : felt, non-woven fabric (NWF) typically used for decoration, and polypropylene, according to Figure 1.

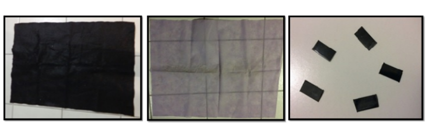

Figure 1: Felt, NWF and pieces of polypropylene, respectively.

These materials, that were easily available for the users all over Brazil, were tested with two very commmom types of floors : laminate and porcelain tiles, using samples of flooring according to Figure 2. The floorings were excited with and without the protective materials. When excited with the protective covering, materials were used unfolded (1 layer) as well as folded two, three and four times.

Figure 2: Tapping machine with the samples of laminate and porcelain tiles, respectively.

Measurements of $L_{n,T}$ and $L_{n,T,w}$ were conducted in the laboratory impact chamber having a solid 10 cm-thick concrete slab. The samples of laminate flooring and porcelain tiles were placed directly on the concrete slab and measurements without and with unfolded and folded proctective coverings were carried out out for different receiver positions.

The polypropylene chips were attached directly to the hammers as proposed by the second method for preparing the MTM according to ISO 10140 :2010 [3] part 5. Felt and NWF were tested according to the first method described in the same standard, placing the material directly on the floor.

3 Results

Both for the reference floor using the STM as well as for each combination of floor/protective covering (MTM) $L_{n,T}$ and $L_{n,T,w}$ were obtained and the influence of the protective coverings were quantified.

3.1 Felt on laminate and porcelain tiles

According to Figure 3, the influence of the felt on the impact SPL measurements with laminate or porcelain titles was very significant, with high values of $\Delta L_{n,T}$, mainly at medium and high frequencies. Changes in $L_{n,T}$ were more evident for the porcelain floor, and increased significantly with the addition of layers by folding the protective covering.

As the standardized impact SPL spectrum was modified using the felt the weighted parameter $L_{n,T,w}$ also does. For the laminate flooring the reductions of $L_{n,T,w}$ were between 1 dB and 5 dB. For porcelain tiles reductions of $L_{n,T,w}$ of 4 dB to 14 dB were found, compared to the standard situation without felt.

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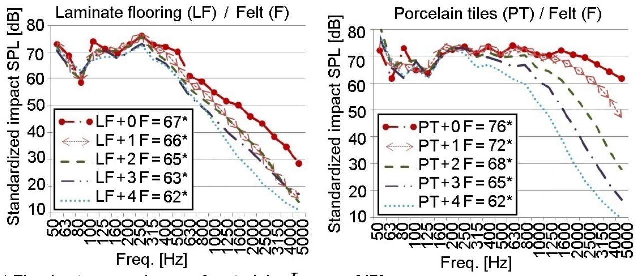
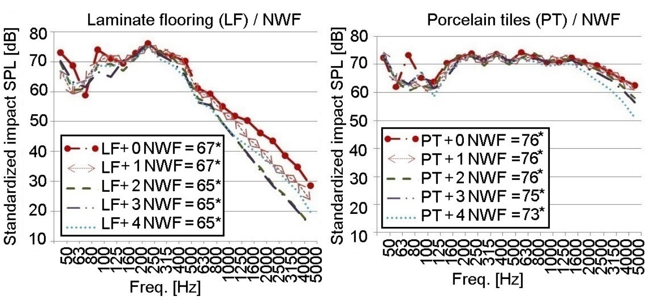


* Flooring type + n. layers of material = $L_{n,T,w}$ [dB]

Figure 3: $L_{n,T}$ and $L_{n,T,w}$ measured without and with n layers of felt on laminate and porcelain tiles.

3.2 NWF on laminate and porcelain tiles

The influence of NWF on the impact SPL measurements wwas found to depend on the type of flooring, according to Figure 4. For laminate flooring, NWF was found to change the $L_{n,T}$ in almost all frequencies, except for some frequency bands between 160 Hz and 630 Hz. On porcelain floors, NWF covering resulted in small reductions of $L_{n,T}$, except for some frequencies, mainly at low and high frequencies.



* Flooring type + n. layers of material = $L_{n,T,w}$ [dB]

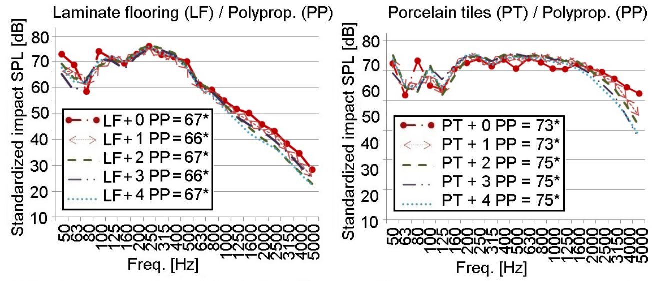
Figure 4: $L_{n,T}$ and $L_{n,T,w}$ measured without and with n layers of NWF on laminate and porcelain tiles.

On laminate flooring, two, three and four layers of NWF changed the $L_{n,T,w}$ up to 2 dB. No reduction in $L_{n,T,w}$ was found for a single layer of NWF. On porcelain tiles, NWF reduced the $L_{n,T,w}$ only when more than three layers were used, resulting in differences from 1 dB to 3 dB compared to the standard situation.

3.3 Polypropylene on laminate and porcelain tiles

Polypropylene chips fixed to the hammers changed $L_{n,T}$ at all frequencies, but more pronounced at low and high frequencies, both for laminate flooring and porcelain tiles, as shown in Figure 5.

On laminate flooring, polypropylene chips on the hammers reduced the $L_{n,T,w}$ by only 1 dB, with one and three layers, and no reduction was found using two and four layers. On porcelain floors, the polypropylene fixed to the hammers resulted in a $L_{n,T,w}$ 2 dB higher than without the protective material, using two, three and four layers.



* Flooring type + n. layers of material = $L_{n,T,w}$ [dB]

Figure 5: $L_{n,T}$ and $L_{n,T,w}$ measured without and with n layers of polypropylene on laminate and porcelain tiles.

4 Discussion

Felt was found to provide the stongest modification of $L_{n,T}$ and $L_{n,T,w}$ compared to other materials (NWF and polipropylene chips), especially on porcelain tiles. Nevertheless it presented an excellent capacity of flooring protection, even with just one layer and no damage to the covering itself from the impact hammers was found.

Using non-woven fabric (NWF) a considerable alteration of $L_{n,T}$ but lower changes in $L_{n,T,w}$ (3 dB on laminate flooring and 2 dB on porcelain tiles) were found. The material was able to protect the flooring only with four or more layers.

Polypropylene was the material that presented minor modifications of $L_{n,T}$ and $L_{n,T,w}$, with alterations of 1 dB on laminate flooring, and 2 dB on porcelain tiles. Is also proved to be a durable material and showed good flooring protection capability.

5 Conclusions

This study demonstrated that polypropylene chips fixed to the hammers ensured proper protection of the flooring and resulted in minor changes to $L_{n,T}$ and $L_{n,T,w}$, for both laminate flooring and porcelain tiles, even with the addition of more layers.

Acknowledgments

The authors would like to thank UFSM for the use of their facilities and equipment and CAPES for a scholarship.

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[2] ISO 140 : 1998 - Acoustics - Measurement of sound insulation in buildings and of building elements. All parts.

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[5] ISO 354 :2003 - Acoustics - Measurement of sound absorption in a reverberation room.

EFFECT OF COMPRESSION ON ACOUSTIC PERFORMANCE OF FIBROUS MATERIALS

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1 Introduction

Fibrous materials are conventionally used in passive applications of noise control and room absorption requirements. When the passive materials are used in HVAC system ducts as well as on the turning vanes of large wind tunnels, the materials compress from their original configurations. Does the compression change the acoustical performance of the porous materials? A series of impedance tube tests were conducted to evaluate the normal absorption coefficients of porous materials under different compressive loads. Preliminary results of the measurement programme are presented in this paper.

2 Background

Porous materials such as rockwool, fibreglass and open-cell foam are usually used in passive absorption applications in HVAC systems and corner-vane treatments in wind tunnels. The acoustic properties of porous materials, such as the absorption coefficient α, vary as a function of its thickness. Larger thickness is needed to provide reasonable absorption at low frequencies. However, the porous materials, in actual systems get compressed under incoming flow. The behaviour of the materials under compression is not well understood. In addition, one is also interested in propagation properties within the materials, such as acoustic density and propagation speed. The above quantities can be related to the flow resistivity of the porous materials [1, 2]. M. Hodgson and his researches at the University of British Columbia have also shown that flow resistivity itself can be a function of frequency [3, 4]. It is important, therefore, to understand the effect of compression on parameters that affect the acoustical performance of porous materials.

3 The Experiment

Two microphone system impedance tubes were applied for the first set of results to evaluate the absorption coefficient α. The impedance tubes are shown in Figure 1 below. The smaller tube is 3.5” in diameter with the frequency range of 350 Hz to 1950 Hz. The larger tube had a diameter of 6.5” with the frequency range of 50 Hz to 1300 Hz. Conventional two-microphone method, using transfer functions, as described by Chung and Blaser was applied to evaluate ‘α’ [5, 6].

The porous materials tested are shown in Figure 2. The height of the materials are: a) 3.5” diameter – Uncompressed height is 5.5” and compressed height is 3.5” and b) 6.5” diameter – Uncompressed height is 6” and compressed height is 3.75”

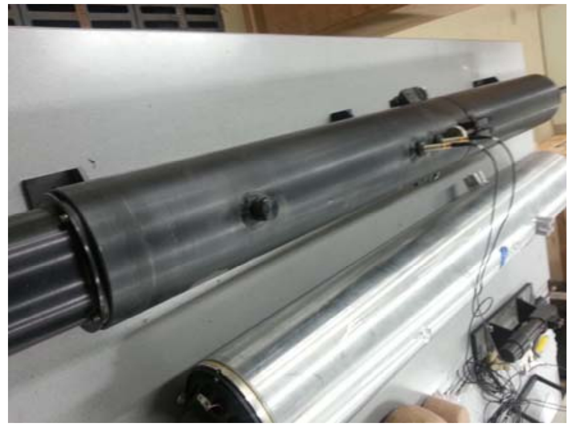

Figure 1: Two circular impedance tubes.

Figure 2: Compressed and uncompressed porous materials.

4 Results and Discussion

A three-microphone impedance tube set-up is shown in Figure 3. The results presented in this paper used only Mic1 and Mic2 to evaluate the absorption coefficient α. A stationary random signal was generated within the impedance tube. A two-channel FFT/Real time analyzer (HP3569A) was used to evaluate the transfer function H_{12} and the microphones were switched to evaluate the reversed transfer function H_{21}. The absorption coefficient is then evaluated from,

$$\alpha = 1 - |R|^2 \quad \textbf{(1)}$$

where,

$$R = \frac{H - e^{-jks}}{e^{jks} - H} e^{j2k(l+s)} \quad \textbf{(2)}$$

R is the complex reflection coefficient, H is the corrected transfer function, s is the spacing between the microphones and l is the distance between the sample and microphone 2 as shown in Figure 3.

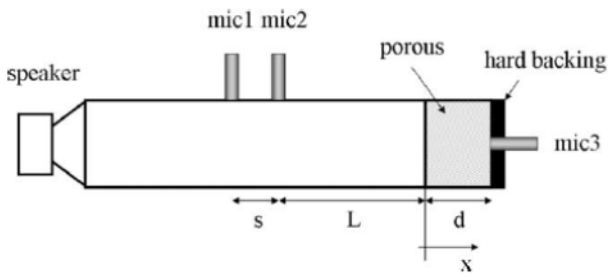

Figure 3: Impedance Tube set-up.

The porous materials shown in Figure 2 were used in the two impedance tubes (3.5: and 6.5" diameter tubes) and α was evaluated. The results are shown in Figures 4 and 5 for one set of samples.

The results of Figures 4 and 5 show the drastic change in α when the material gets compressed. The effect seems to be predominant at low frequencies between 50 and 250 Hz. Even in mid frequency bands α is seen to decrease, albeit, at a smaller rate. For example, α reduced a) from 0.41 to 0.21 at 100 Hz; b) from 0.85 to 0.49 at 175 Hz; c) from 0.88 to 0.65 at 200 Hz; and d) 0.85 to 0.76 at 650 Hz.

Thick porous materials of the order of 12" are used to treat the large turning vanes in a full scale wind tunnel, so as to achieve good performance at low frequencies. The above results show that the performance can decrease by as much as by 50% when the materials get compressed by the impinging flow.

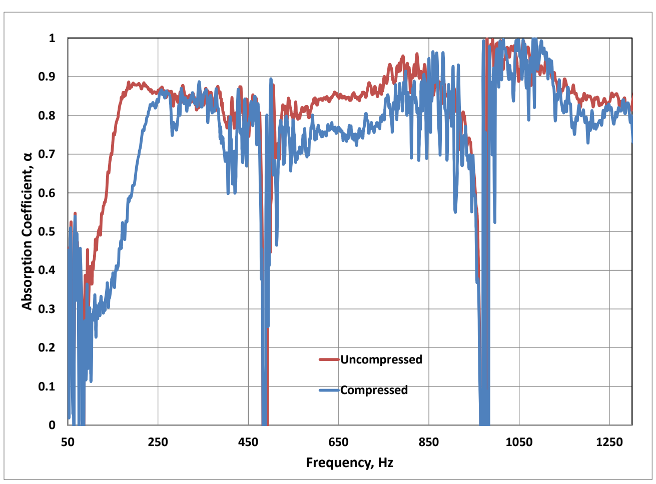

Figure 4: Absorption coefficient of porous materials (6.5" tube)

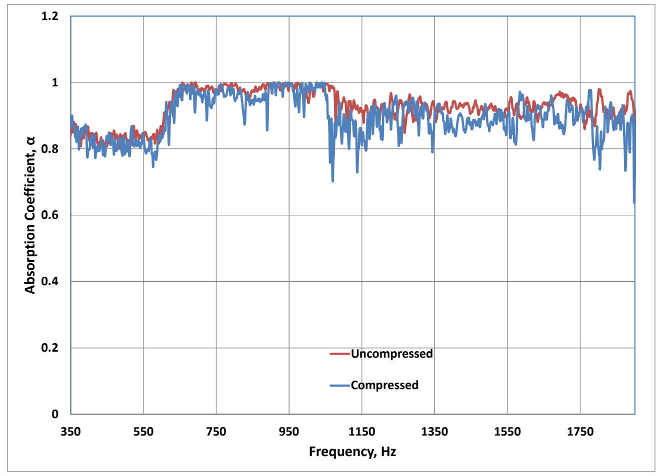

Figure 5: Absorption coefficient of porous materials (3.5" tube)

5 Conclusion

Absorption coefficient α of porous materials was measured by using a two-microphone impedance system. The results showed that α gets degraded at low frequencies when the materials gets compressed.

Acknowledgments

We would like to acknowledge that the work was conducted under a research grant from the Ontario Centre of Excellence with Roxul Inc. as the industry partner. We would like to thank Prof. Hodgson and his research group for at the University of British Columbia for providing performance results of porous materials. We would also like to acknowledge Aiolos Engineering Corporation for the loan of four impedance tubes of different lengths and sizes.

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ENGINEERING ACOUSTICS / NOISE CONTROL - INGÉNIERIE ACOUSTIQUE / CONTRÔLE DU BRUIT

ROOFTOP NOISE IMPACT INVESTIGATION TO THE COMMUNITY

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1 Introduction

Based on an investigation of rooftop equipment noise, the methodology described in this paper was developed to isolate the noise contribution from a subset of noise sources from a set of noise sources (including urban hum) that contribute to the sound pressure level at a given community receptor location. Results comparing the projected noise impact from the rooftop equipment to community measurements based on this method are presented.

2 Methodology

The following parameters are considered in developing the methodology:

1. The modeled partial sound pressure level contribution for each source; and

2. The measured sound pressure levels at Reference Receptor locations that surround the sources of interest.

The term "Reference Receptor" refers to the noise monitor location that captures specific sources of interest. Typically, multiple reference receptors are used to surround the area of interest, and are located between the sources and community receptor but comparatively closer to the sources, such that the reference receptor is not influenced by extraneous noise such as urban hum.

Figure 1 presents a graphic overview of this method in terms of the main relationships between sources, reference receptors and community receptors.

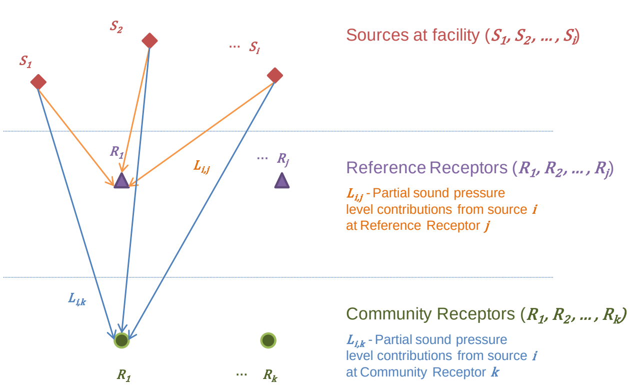

Figure 1: Relationship between Sources, Reference Receptors and Community Receptors

In terms of the method the attenuation that occurs during propagation is considered constant. It is not dependent on the source sound power level, but is related on factors such as a) source-receptor geometry including heights and elevations; b) attenuation due to geometrical divergence; c) screening effects of the building and surrounding topography; d) attenuation due to atmospheric absorption; e) attenuation due to ground effects; and f) meteorological correction factors favourable for propagation from the sound source to the receptors. This means that any increase or decrease of the source sound power level will have a direct and proportional effect on the reference and community receptor partial sound pressure level. This is better illustrated in Figure 2, where a single source, multiple reference receptor and single community receptor is presented.

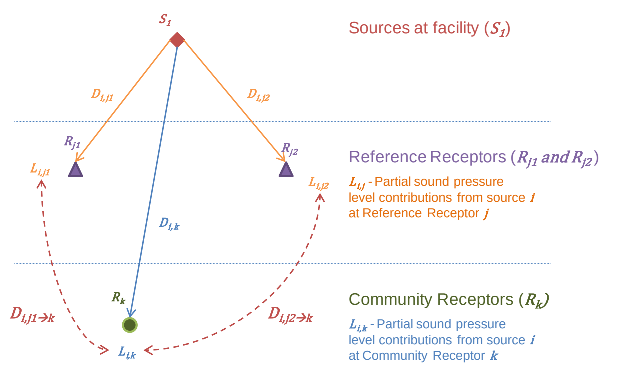

Figure 2: Single Source Overview of Relationship between Partial Sound Pressure Contribution, Source Sound Power and Attenuation Factor

Acoustic modeling provides the partial sound pressure level contribution from each source at community and reference receptor locations, and provides the relative contribution from each source at a given reference receptor location j. Assuming that the urban hum contribution at the reference receptor location is insignificant, any variations of the sound pressure level at the reference locations is then dependant only on variations of the sound power level of the sources contribution to the reference receptor. The approach taken here assumes the following:

a) All the sources identified from the facility that are captured by the reference receptor are continuously operating; and

b) The background noise levels due to urban hum at the reference receptor location are negligible.

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The following equation can be written when noise level contributions are considered in terms of sound pressure in Pa (instead of sound pressure level in dB):

$$\frac{p_{i,j}^{2}{}'}{p_0^2} = \langle p'_{i,j} \rangle = 10^{\frac{L'_{i,j}}{10}} \textbf{ [1]}$$

And the total sound pressure level at reference monitor location *j* becomes:

$$\langle p'_j \rangle = 10^{\frac{L'_j}{10}} = \sum_{i=1}^{\#Sources} \langle p'_{i,j} \rangle \textbf{ [2]}$$

From partial $\langle p'_{i,j} \rangle$ and the total $\langle p'_j \rangle$ sound pressure, apparent weighting for each partial sound pressure to the total sound pressure can be determined from the formula:

$$X'_{i,j} = \frac{\langle p'_{i,j} \rangle}{\langle p'_j \rangle} \textbf{ [3]}$$

The following is assumed in developing this methodology:

1. The apparent weighting factors (Equation 3) from the model are transferable to the measured values at the Reference Receptor Locations.

2. The method provides long term average sound levels. As such any differences between measured total sound pressure levels at the Reference Receptor Location *j* (L_j) and modeled total sound pressure levels at the Reference Receptor Location *j* (L'_j) can be attributed only to the sources. Since all the sources are considered to be continuously operating, the difference in sound pressure level is distributed according to the apparent weighting.

For each source *i* under consideration and based on the assumptions above, multiple partial sound pressure level contributions $L_{i,j \to k}$, based on different Reference Receptor Location *j*, should provide similar values of $L_{i,j \to k}$.

In reality, different sources of error in both measurements and varying background sound pressure level may lead to slightly different partial sound pressure level contribution $L_{i,j \to k}$ than provided in the acoustic modeling. Multiple sound pressure level contributions from various Reference Receptor Locations $L_{i,j \to k}$ are logarithmically averaged for each source to adjust for this.

$$\overline{L_{i,k}} = 10 \times \log_{10}\left(\frac{1}{\#j} \times \sum_{j=1}^{\#j} 10^{\frac{L_{i,j \to k}}{10}}\right) \textbf{ [4]}$$

From the individual average source contributions $\overline{L_{i,k}}$ the total sound pressure level at Community Receptor Location *k*, can be determined by adding the individual source sound pressure level contributions at the receptor.

$$L_k = 10 \times \log_{10}\left(\sum_{i=1}^{\#S} 10^{\frac{\overline{L_{i,k}}}{10}}\right) \textbf{ [5]}$$

3 Results

Rooftop equipment on a facility was surrounded with four noise monitors, and compared to measurements at three community locations. To determine rooftop noise contribution to the community monitor locations, the measured rooftop noise levels were projected using the methodology described above to the community monitor locations on a 1-sec Leq basis. A sample one hour comparison is shown in Figure 3.

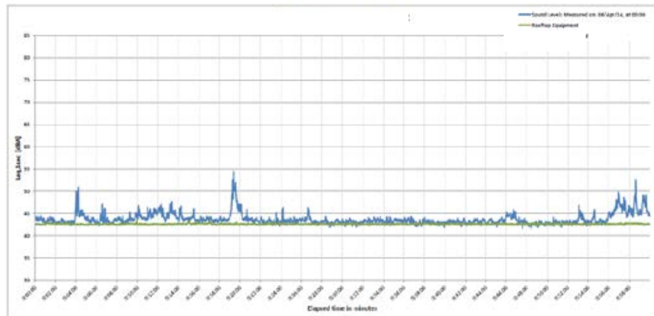

Figure 3: Rooftop Equipment Projected Noise Levels at the community vs Community Measured Noise Levels

Comparison over the long term monitoring at the community locations showed a good correlation to the community measurements. In this case, rooftop noise was typically steady state, and did not contribute to peak noise levels at the community as confirmed by audio recordings.

4 Conclusion

By comparing projected noise and measured community noise, the following conclusions were drawn: 1) rooftop noise is within acceptable noise guidelines, 2) peak sound levels experienced in the community were not contributed by the rooftop equipment, and 3) the rooftop equipment is a contributing factor to the ambient noise in the community.

This methodology was developed to project facility noise to the community and isolating it from urban hum. The methodology outlined in this paper allows for the reference monitoring around a set of noise sources, and projecting the noise levels from these reference monitors.

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TRANSIENT RADIATION BY A SYSTEM OF TWO CYLINDRICAL SHELLS

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1. INTRODUCTION

Although a system of two co-axial cylindrical shells has been considered before in both shock and acoustic loading contexts [1, 2], there is very limited information available in the existing literature regarding the structure of the acoustic field radiated by such a system in cases when the loading is transient in nature. Therefore, our goal is to offer insights into the dynamics and composition of the field for a transient external acoustic loading.

2 MATHEMATICAL FORMULATION AND SOLUTION METHODOLOGY

We employ the semi-analytical methodology that we have developed in recent years [3-7] while accounting for the fact that now there are two structures, one interacting with both external and inter-shell fluids, and one interacting only with the inter-shell fluid.

More specifically, the fluids are modelled using the wave equation, and the shells are modelled using the Kirchhoff-Love linear theory of thin shells. The analytical part of the solution employs the separation of variables and Laplace transform with respect to the spatial and temporal coordinates, respectively, and the fluid and structural parts are then coupled using a 1D finite-difference approach.

The resulting solution possesses the advantages of being very computationally efficient and robust, while its main limitation is the restriction on the complexity of system's geometry in that it can only handle cylindrical structures.

3 RESULTS AND DISCUSSION

We considered a system of two steel shells with the radius of the outer shell of 1 m, the radius of the inner shell 0.5 m, and the thickness-to-radius ratios of the shells being 0.01 and 0.02, respectively. Both the outer and the inter-shell fluids are assumed to be water. The incident acoustic load is assumed to be a single point-source pulse originated at the stand-off distance of four radii of the outer shell and with the exponential decay of the pressure behind the front, the decay constant being 0.0001314 s.

We first address the acoustic patterns observed in the system, and Figure 1 shows two representative snapshots of the acoustic field induced by the load in question. One of the immediately apparent features is the dramatic effect that the inner shell has on the overall acoustic pattern radiated by the system - the external load, upon its transition into the

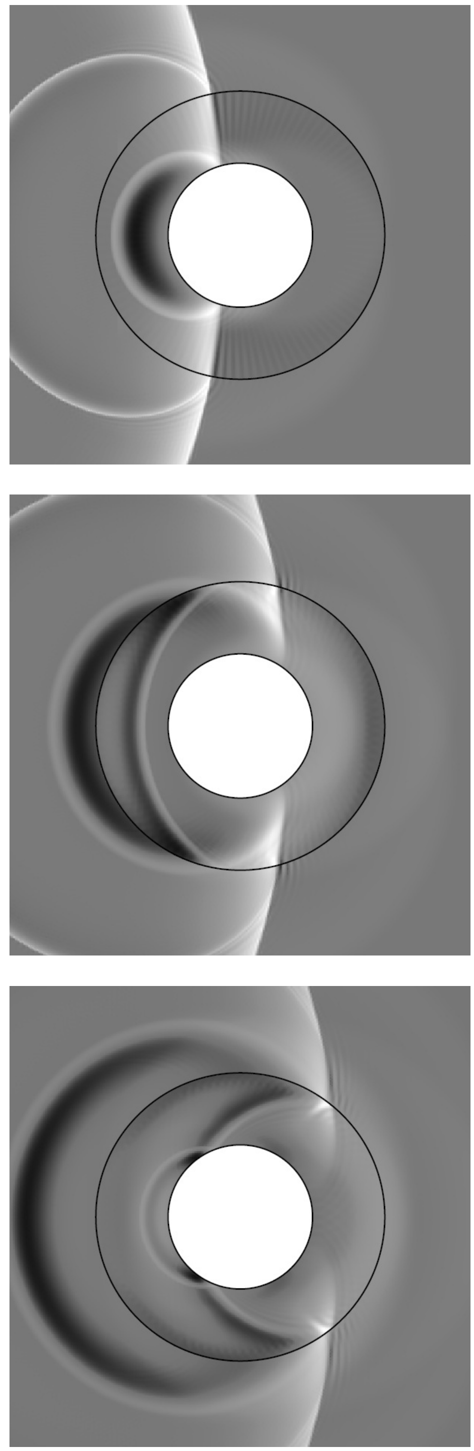

Figure 1. Representative snapshots of the acoustic field.

inter-shell fluid, reflects off the inner shell and then propagates upstream, reaching the outer shell and reflecting off it again, with the reflected wave propagating downstream, and with another wave originated in the external fluid that still propagates upstream, thus constituting an effect that might be referred to as the "leaking" of the waves propagating in the inter-shell volume into the external fluid domain.

We further remark that as the reflected wave propagates downstream, the just described process of reflection from the inner shell is repeated, thus demonstrating that multiple reflections of the waves in the inter-shell volume are important contributors not only to the internal but also to the external acoustic field observed during the interaction.

4 CONCLUSIONS

We have employed a semi-analytical model of non-stationary fluid-structure interaction that was developed with the main goal of simulating the acoustic field radiated by a submerged system of two cylindrical shells. The model allowed us to address a typical two-shell configuration, and both shells were seen to have a significant effect on the overall structure of the field radiated by the system.

ACKNOWLEDGEMENTS

The present study was funded by the Defence Research and Development Canada (DRDC) Atlantic. The authors also wish to express their gratitude to Dr. Malcolm Smith for his advice and many invaluable comments and insights that he was always willing to provide and that to a large degree shaped the course of this work.

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MODELING ACOUSTIC RESPONSE OF STRUCTURALLY COMPLEX SHELL SYSTEMS

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1. INTRODUCTION

One possible approach to modeling non-stationary acoustic radiation by cylindrical shells is to employ the classical analytical methodology of mathematical physics (e.g. [1-4]). The resulting methodology (which we will refer to as the "response-functions-based methodology") is very robust and computationally efficient, but, unfortunately, it has a very substantial shortcoming of not being able to handle geometries different from the simplest classical ones (e.g. spherical and cylindrical). In order to overcome this lack of versatility, an idea was proposed to combine the response-functions-based approach with FEM, thus producing a methodology that would combine the computational efficiency of the former with the geometrical versatility of the latter.

2 Problem formulation and solution methodology

The fluid domain is assumed to be governed by the wave equation, which is solved using a combination of the separation of variables and Laplace transform (spatial and temporal variables, respectively). The methodology has been discussed in detail in our earlier work [1-6], and we do not reproduce the respective discussion here, but summarize that the approach produces the acoustic components (diffracted and radiated one) in the form of a Fourier series with time-dependent coefficients which, for the radiated pressure, also depend on the normal displacements of the outer surface of the shell.

The structural part is approached using a 2D plane strain model discretized with the finite element methodology, with the normal displacements of the structural surface passed to the fluid solver at each time step and the total acoustic pressure received from the fluid solver, also at each time step, thus the coupling between the two solvers is accomplished.

3 RESULTS AND DISCUSSION

In order to test the capabilities of the methodology, we considered several structural configurations, and present the results for one of them here. But first, we remark that the model has been extensively tested using several benchmark problems, and a very good agreement with experimental data was observed for thin shells without any structural enhancements.

One of the most typical scenarios of industrial relevance is a shell with two attached masses of dimensions that are comparable with the thickness of the shell, Figure 1.

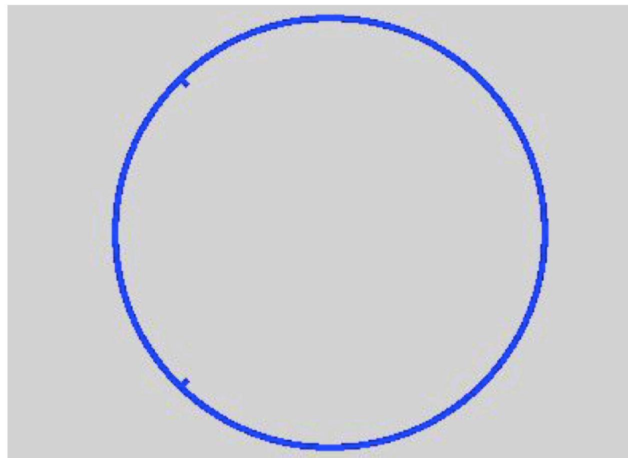

Figure 1. Shell with two "light" attached masses.

For such a configuration, the attached masses were observed to act in a "source-like" manner when the structure (a steel shell with the thickness-to-radius ratio of 0.03 submerged into water) was subjected to a single acoustic pulse, Figure 2. This behavior is a very characteristic feature that was observed in similar systems in experimental studies that offer acoustic field images [7], and having demonstrated that this behavior indeed takes place provides additional confidence in the physical adequacy of the developed model.

We note that the respective wave pattern is rather complex, and appears to have several different components to it. This is also consistent with the experimental results [7] where several different types of radiated waves were seen to emerge, in particular the symmetric Lamb wave S_0, the antisymmetric (or pseudo-Rayleigh) wave A_0, and the Scholte-Stoneley wave A. Thus, the present approach appears to be quite adequate for accurate modeling of the complete structure of the radiated field since the fluid solver has been demonstrated to produce very good results for simple shells in that it accurately reproduces all known types of radiated waves [8].

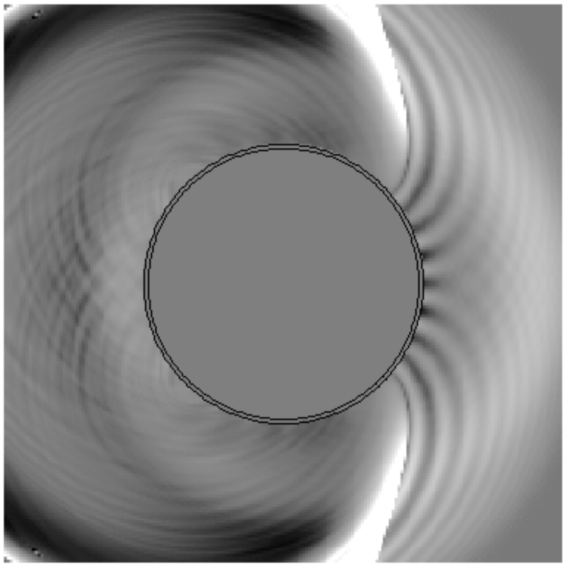

Figure 2. The "source-like" behavior of the attached mass.

6 CONCLUSIONS

Judging from a number of studies that we carried out for some of the most typical structural configurations of industrial relevance, we can conclude that the proposed methodology appears to be a very efficient tool for modeling the acoustic response of cylindrically shaped, structurally complex submerged structures.

The methodology appears to be particularly attractive for the use at the pre-design stage where it is often necessary to carry out extensive parametric studies, and where higher computational efficiency is a considerable asset.

ACKNOWLEDGEMENTS

S.I. gratefully acknowledges the support of the Natural Sciences and Engineering Research Council (NSERC) of Canada. The mobility grant for scholars provided by the Erasmus Mundus program in the framework of the Master of Science in Computational Mechanics (FPA 2013-0220 2014 Erasmus+) is also acknowledged.

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Comparison of Airborne and Impact Sound Insulation Improvement due to Adding Toppings on Homogeneous Floors

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1 Introduction

For the upcoming change of the airborne sound insulation requirement in the National Building Code [1], measuring direct and flanking path data is one way to predict the ASTC rating of the system [2] . In order to minimize the overlap of measured airborne and impact floor data, NRC has looked at interpolating floor topping improvements to similar floors type. In this paper, homogeneous floors are considered.

The first part of the results section will compare the one-third octave airborne and light impact improvements due to toppings on cross-laminated timber (CLT) and concrete floors. The second part will focus on the improvement of the single number ratings (SNR) of these floor types.

2 Specimen Descriptions and Properties

Each floor is described by a short code of the elements it is composed of, starting from the top to the bottom layer. The number following the short code is the thickness of the layer in mm. For example, CON38_RESL9_CON200 describes a 38 mm concrete topping installed on a 9 mm resilient interlayer on top of a 200 mm concrete slab. There are three base floor assemblies that are compared in this paper :

— CLT5 - 5 ply - 175 mm thick
area density = 91 kg/m^2 (525 kg/m^3)
— CLT7 - 7 ply - 245 mm thick
area density = 130 kg/m^2(525 kg/m^3)
— CON 200 - 200 mm thick
area density = 504 kg/m^2(2520 kg/m^3)

with one topping :

— CON 38 - 38 mm thick - 103 kg/m^2

installed on two different resilient mats :

— RESL8 - 8 mm thick rubber - 3.6 kg/m^2
— RESL9 - 9 mm thick closed-cell foam - 0.3 kg/m^2

Each assembly was tested according to ASTM E90 and ASTM E492 in the NRC floor facility which has a test opening of 17.9 m^2 and room volumes of 175 m^3 each.

3 Results and Discussion

3.1 Third Octave Band Comparison

Plotting all airborne TL curves in Figure 1 shows improvement in TL due to the addition of a topping on a bare CLT5 floor (up to 12 STC points). The ΔTL of the topped floor assemblies (Figure 2) show 5 to 10 dB more improvement for the same topping and interlayer on the CLT5 floor, compared to the CON200 floor.

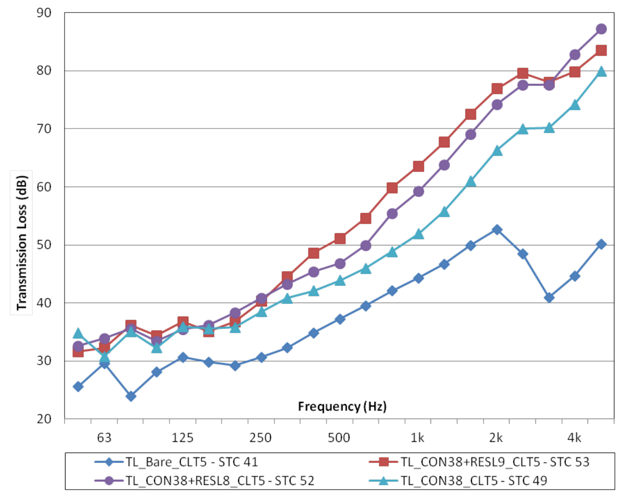

Figure 1: TL data of CLT5 floor with various toppings

One approach to reduce airborne sound transmission through a partition is to increase the mass of that partition. For example, installing a 103 kg/m^2 concrete topping on a 91 kg/m^2 base CLT more than doubles the mass of the base assembly. By doubling the mass, there is a theoretical 6 dB increase to the TL curve, which is observed up to 200 Hz (see Figure 2). Above the mass-spring resonance at 200 Hz, there is additional losses due to decoupling of the floor elements.

Figure 2: ΔTL data of CLT 5 and CON 200 with various toppings

Plotting the improvement of impact sound pressure levels (ΔISPL) of the same six floor assemblies (Figure 3), the opposite trend of the ΔTL can be observed : the floor toppings reduce the impact noise more on the concrete floor

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than on the CLT5 floor. The addition of a decoupled topping onto the bare concrete floor reduces the mechanical impedance match of the steel impact hammers and the concrete floor, thus reducing the power injected into the floor.

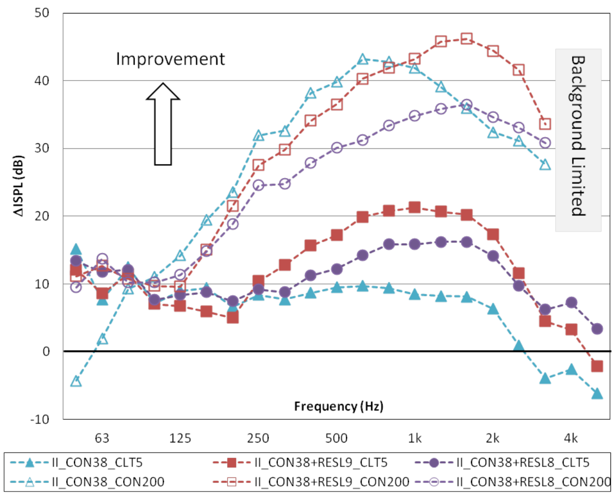

Figure 3: ΔISPL data of CLT 5 vs CON 200 with various toppings

The differences of the ΔTLs and ΔISPLs, found in Figure 2 and 3, are plotted in Figure 4. They show the same tendencies for floors with similar base assembly (either CLT5 or CON200).

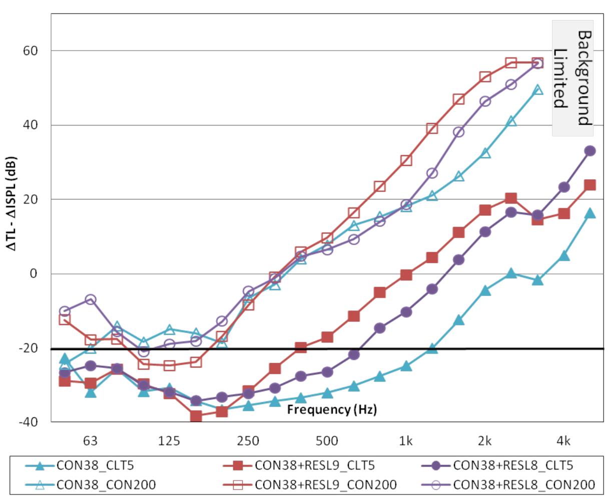

Figure 4: ΔTL - ΔISPL (CLT5 and CON200)

Looking at the ΔTL and ΔISPL of the CON38_RESL9 topping on three base floors (Figure 5), the CLT floors have the same trends for both ΔTL and ΔISPL up to 1600 Hz. For the CON200 floor, the ISPL improvement is much greater than for the TL.

3.2 Single Number Ratings

The single number ratings of the bare floor assemblies with the Δ improvement calculated using only the SNRs (not the E2179 method) of the toppings are found in Table 1. The addition of the CON38_RESL9 on the CLT floors offers very comparable increase for both the TL and ISPL (+12 vs. +11 and +13 vs. +11), whereas for the concrete slab, only the IIC rating is significantly boosted.

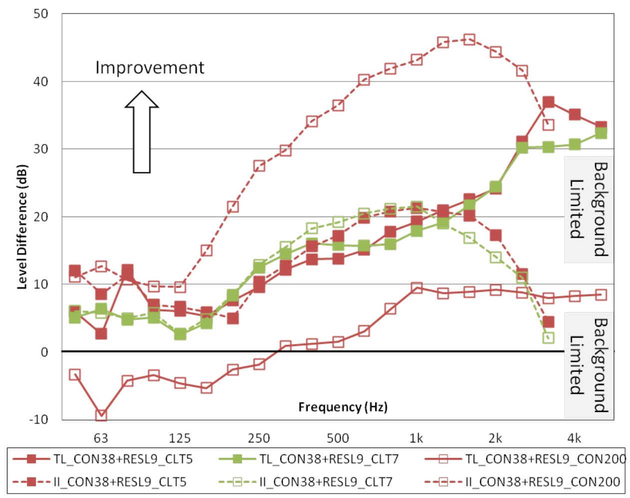

Figure 5: ΔTL and ΔISPL data of CON38_RESL9

Table 1: Single number ratings and improvements

	CLT 5		CLT 7		CON 200	
	STC	IIC	STC	IIC	STC	IIC
Bare	41	25	44	30	57	30
Δ Improvement						
CON38_RESL9	+12	+11	+13	+11	-1	+29

4 Conclusions

CLT floors of varying thickness react very similarly to the same topping. The addition of a concrete topping on a CLT and a concrete floor have different effects for both airborne and impact sources, and the one-third octave and SNR improvements cannot be transferred between the assemblies. In the near future, NRC is hoping to gather more data on other types of floors (steel joists, precast concrete, concrete), in order to populate the topping database and find similarities where they might exist. Currently, there is no ASTM standard method to easily calculate the ΔSTC of homogeneous floors. For the ΔIIC of CLT floors, it might be possible to define a reference IIC curve for CLT floors that might be used with the ASTM E2179 standard.

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DETERMINING TRAIN VELOCITY FROM SOUND LEVEL MEASUREMENTS

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1 Introduction

In environmental noise studies, unattended, long-term sound level measurements are common practice when determining sound emission from rail sources. In many cases, knowing the train velocity during the measurement can add significant value. The velocity may help to further characterize the source and allow for comparisons between modelled and measured results. However, during unattended measurements observing the train velocity directly is not possible.

This paper presents a means of determining train passby velocity from a time history sound level measurement. This method may be used when conducting unattended sound level measurements.

2 Methodology

2.1 Measurements

Sound level measurements of passenger train passbys were conducted in conjunction with velocity measurements. Overall sound pressure levels were recorded every 0.5 seconds in A-weighted decibels. The sound level meter was located at approximately 1.5 metres above grade at a distance of 13.6 metres from a straight section of railway. The full time history measurement for a single passby was approximately 3 to 4 minutes in duration. Measurements were started when the train was first audible over ambient sound levels and terminated when the train was no longer audible over ambient sound levels.

A typical time history measurement of a train passby can be split into three segments:

1. locomotive approach – characterized by steadily rising sound levels over time;
2. locomotive retreat – characterized by steadily declining sound levels with time; and
3. car passbys – characterized by relatively constant sound levels which are, in an ideal measurement, more than 20 dB above the ambient sound level.

These segments are presented in Figure 1. The locomotive is generally 20 dB louder than the rail car passby. One must review the recorded time history to ensure that there is at least 10 dB difference between the locomotive and rail car passby samples because the prediction method requires an uncontaminated locomotive approach and retreat samples. A 20 dB difference between the locomotive and cars can be considered uncontaminated, a 10 dB difference may have slight, yet acceptable, amounts of contamination.

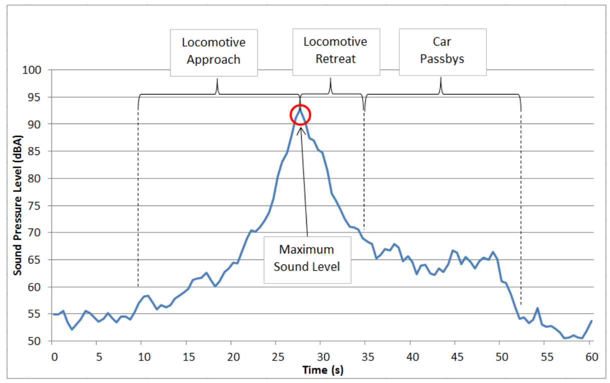

Figure 1: Time history measurements of passenger train passby

The maximum sound level occurs between the locomotive approach and retreat and when the train is at the minimum distance between the track and measurement location (perpendicular distance).

For determining train velocity, the relevant segments of the measurement will be the locomotive approach and retreat.

2.2 Assumptions

The calculation method requires some assumptions to be made about the locomotive and track geometry.

The calculation method assumes that the train is operating on a straight piece of track. As such, when employing this technique, the measurement location should be carefully selected such that there is a significant stretch of straight track on either side of the measurement location.

The train locomotive is assumed to be a single moving point source with an omni-directional emission directivity. The moving point source is assumed to be dominant over the wheel noise from the other train cars during approach and retreat. This assumption is especially valid during the train approach where the trailing cars are further from the measurement location than the locomotive.

The train is assumed to be moving at a constant velocity with the engine under constant load. The sound emission from the engine is assumed to remain constant throughout the passby. However, the measured sound level will vary thoughout the passby as the distance between source and measurement point decreases and increases.

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2.3 Calculations

The calculation method requires that sound pressure level measurements be conducted at constant time intervals and at a known distance from the rail line; this type of measurement method is often referred to as a 'time history' measurement. As the train passes the sound level meter records sound pressure level measurements, which correspond to the various train positions on the track. The maximum sound pressure level recorded, assuming an omni-directional source, occurs when the train is at the perpendicular distance between the rail line and measurement point.

The calculation uses a sound pressure level distance propagation formula:

$$d_n = d_0 \cdot 10^{\left(\frac{SPL_0 - SPL_n}{20}\right)}$$

Where SPL_n is the measured sound pressure level for a given time history measurement, and d_n is the distance, in metres, between the source and measurement location. The SPL_0 and d_0 represent the maximum recorded SPL and the perpendicular distance from the measurement distance to the rail line. These variables are presented in Figure 2.

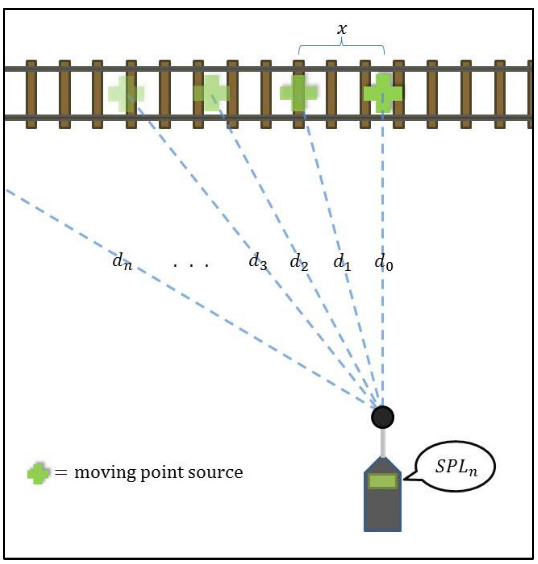

Figure 2: Relevant variables in calculation

Solving for d_n allows us to solve for x, the distance travelled during each time history interval. The train velocity is determined by dividing x by the time history interval (t).

$$velocity = \frac{x}{t} = \frac{\sqrt{{d_n}^2 - {d_0}^2}}{t}$$

3 Results & Discussion

Using the recorded measurements, the train velocity was calculated to be approximately ±10% of the measured train velocity. To visually compare the moving point source calculation to the measured sound data, the equation was re-arranged to calculate the sound level for a given velocity. This comparison, presented in Figure 3, showed good agreement near the point of maximum sound level, but poor agreement where the locomotive was further from the measurement point.

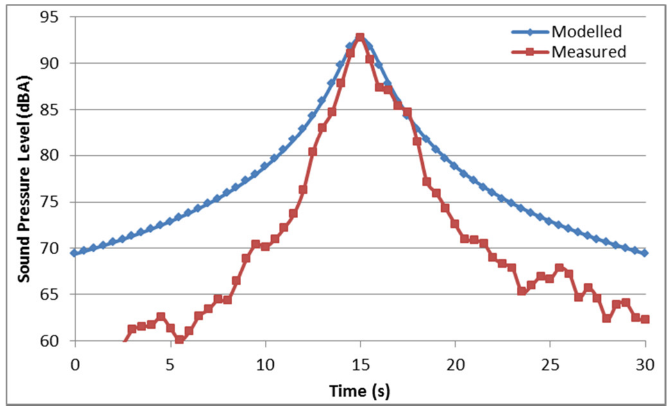

Figure 3: Comparison of modelled and measured passbys

Potential sources of error in the prediction method may arise where a given assumption is not necessarily valid. The track geometry, source characteristics and operating conditions of the train may affect the calculated train velocity.

The calculation method assumes that the train is operating on a straight section of track. Where monitoring is conducted on non-straight tracks the calculation for x may need to be modified appropriately. Care should be taken to select a monitoring location with an appropriate amount of straight track on either side of microphone.

The train locomotive is assumed to be a single moving point source with an omni-directional emission directivity. However, no field studies or literature review have been conducted by RWDI to assess validity of this assumption. Further measurements to characterize different types of locomotives (commuter, freight, etc…) may improve results from the prediction methodology.

As the measurements are generally unattended it is not always possible to identify if the is train operating at a constant velocity or with the engine under constant load. Where the train is accelerating or operating under varying loads the data may not yield an accurate velocity prediction, or the data may not be useful for this method at all. If monitoring locations are located in close proximity to stations where the train would stop, effects of acceleration or deceleration should be considered.

4 Conclusion

The velocity of passenger trains was calculated to within ±10% of the measured velocity through analysis of time history measurements of sound pressure level. This method could be used to determine the velocity of train passbys for unattended sound level measurements.

MEASUREMENT OF STRUCTURAL REVERBERATION TIMES FOR CALCULATION OF ASTC IN UPCOMING NBCC

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1 Introduction

The 2015 edition of the National Building Code of Canada (NBCC) will likely see a change from acoustic requirements of element performance to system performance. It is anticipated that there will be three paths for compliance : field measurements in the finished building, look-up tables with representative assemblies, and a detailed design procedure using laboratory test data and prediction methods. For monolithic construction such as concrete or concrete masonry buildings, the prediction is based mainly on ISO 15712-1 [1]. In this international standard, the structural reverberation time of building elements is an important input parameter for the transfer of laboratory data to *in-situ* values.

The measurement of structural reverberation times (RTs) is similar but more difficult than the measurement of room reverberation times. ISO 10848-1 [2] describes the procedure, and more comprehensive descriptions can be found in the literature [3,4]. For this paper, the measurement of structural RTs was investigated in two case studies, for a 190 mm concrete block wall and for a 203 mm hollow-core concrete floor. Due to space constraints, only the floor measurements are presented here. Common challenges are highlighted, and various options regarding the measurement and evaluation of structural impulse responses are investigated.

2 Determination of structural reverberation times by backwards integration

The most common method of determining structural RTs uses backwards-integration of impulse responses [5]. Impulse responses are obtained by either exciting the structure under test with a hammer and recording the response directly, or by exciting the structure using a shaker with a sweep or noise signal, and subsequent signal processing. ISO 10848-1 gives guidance on the number and location of excitation and response positions.

Once the impulse responses have been obtained, a series of signal processing steps are performed to estimate the structural RTs. The impulse response is filtered into (octave or third-octave) frequency bands, and an exponential time window is applied. At this point, it is recommended but not required to remove the noise floor in the impulse responses [6]. In a last step, the energy decay curves (EDCs) are calculated by backwards-integration of the squared impulse response, and the RTs are estimated by linear regression.

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3 Measurement setup

Measurements were performed on a 203 mm thick hollow-core concrete floor of size 4.70 m×3.80 m, installed in the floor testing facility at NRC. The impulse responses were obtained using two different shakers, a small Wilcoxon F3 shaker and a large PCB K2007E01 shaker, driven with an exponential sweep of length 40 s, and also using two impact hammers. Three excitation positions were used, and four different response positions each. The location of the positions was according to ISO 10848-1. For the hammer measurements, three repeats were recorded for each excitation point, and as required by ISO 10848-1 different hammer tips were used in order to excite a wider frequency range.

The responses were recorded using accelerometers type PCB 320C03 with a sensitivity of 1 mV/m/s^{-2}. The sensors were magnetically connected to metal tabs which were epoxied to the surface of the floor specimen. The small shaker was threaded directly to a metal tab which was also epoxied. The larger shaker required to be supported and, in order to prevent rotational excitation, it was connected using a flexible stinger reinforced with a metal sleeve. An impedance head (PCB 288D01) monitored the input force and acceleration.

4 Evaluation of the impulse responses

The measured impulse responses were evaluated using an in-house software tool implemented in MATLAB. The software calculates the EDCs using the procedure described above, and estimates the RTs using a default evaluation range between -5 dB and -15 dB. It rejects invalid estimates based on a minimum signal-to-noise ratio (>25 dB), correlation coefficient between EDC and regression line (>95%), and BT product (>8 for forward filtering, >4 for reverse filtering). The regression lines can be adjusted manually. Given the large amount of data (# of EDCs = # of excitation positions × # of response positions × # of repeats × # of frequency bands), manual adjustment is very time-consuming.

Several evaluation parameters were investigated :

- Noise truncation limits the impact of background noise on the determined RTs [6].
- Traditional forward-filtering limits the lowest measurable RT more severely than reverse filtering [7]. The parameter of interest is the product of filter bandwidth B and RT T (BT product).

5 Results

The effect of forward and reverse band filtering is considered first. Figure 1 shows the mean values of the determined RTs for the hollow-core floor, measured with the large shaker. Also shown are the limits for the lowest measurable RTs. For this case study, the type of filtering did not have a significant effect, as the structural RTs were above the limits of both forward and reverse filtering. For structures with higher losses these limits will come into play, and therefore the BT products of the RTs should always be checked. If they are below the limit, reverse filtering should be used, or the frequency resolution should be decreased (octave bands instead of third-octave bands).

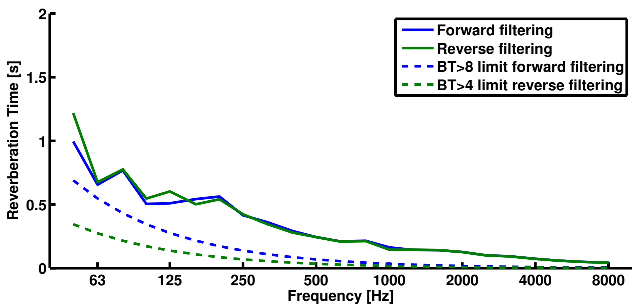

Figure 1: Comparison of forward and reverse frequency band filtering, for hollow-core floor measurements using the large shaker

The influence of noise truncation is considered next. Figure 2 shows the RTs of the hollow-core floor, this time measured with the small shaker. The initial RT estimate (blue line) shows considerable deviations below 400 Hz. Manual adjustment of the fitted regression lines to the EDCs (green line) removes the outliers and yields acceptable standard deviations. However, compared with the automated results with enabled noise truncation (red line), the values are still systematically higher at low frequencies. This is because the noise floor in the impulse responses is integrated in the calculation of the EDCs, resulting in skewed values for the RTs. Manual adjustment of the red line (not shown) had only very little effect on the RT values. From Figure 2 it is concluded that the use of some noise removal algorithm can be highly beneficial. However, it should also be noted that the effect of noise truncation was significantly less pronounced for the large shaker. The reason is that the signal-to-noise ratio for the large shaker was much higher, so noise in the impulse responses had less influence. The small shaker is not strong enough to excite the specimen properly at low frequencies.

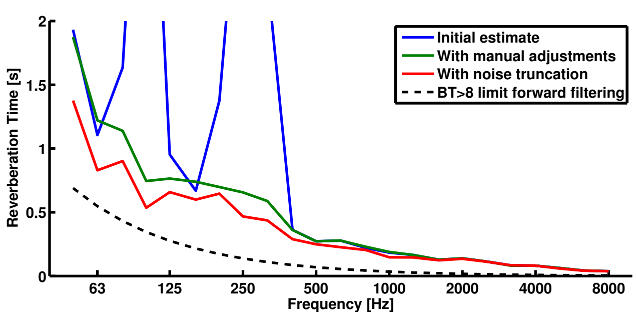

Figure 2: Effect of noise truncation for hollow-core floor measurements, using the small shaker

Finally, the influence of the type of excitation is considered. Figure 3 shows the RTs of the hollow-core floor, measured with different excitation sources. Reverse filtering was used, noise truncation was applied, and the regression lines were manually adjusted. Despite these provisions, the RTs deviate considerably below 500 Hz, in particular between shaker and hammer measurements. This discrepancy has been observed before in other laboratories [8]. One reason may be found in the significant difference in exerted force, pointing to possible non-linear behaviour of the structure under test. More research is required on this issue.

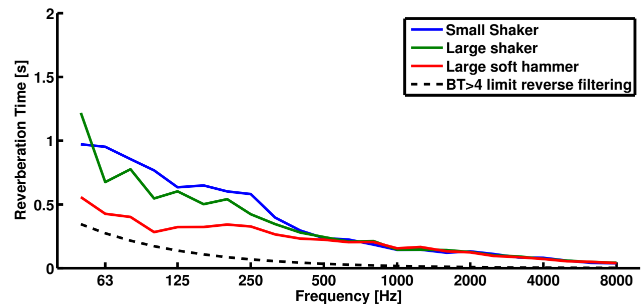

Figure 3: Comparison of RTs obtained with different excitation sources, for concrete hollow-core floor

6 Conclusions

In the cases studied, it was found that there can be systematic differences between RTs obtained with hammers and shakers, respectively. This confirms previous research [8]. Until more research has been conducted on these deviations, it is recommended to use a shaker, because the forces exerted by the shaker closer resemble the forces exerted by airborne excitation. Furthermore, it is recommended that noise truncation techniques [6] be used, as they can significantly improve the RT estimates. In the cases studied, the differences between forward or reverse filtering were not significant. For structures with higher losses it is recommended to use reverse filtering.

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PARAMETRIC ANALYSIS OF ELBOW SILENCERS

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1 Introduction

Porous materials are usually applied in HVAC system ducts as passive silencers. The application of porous as passive absorbers in elbow silencers has been investigated. The performance of the elbow silencers is expressed in terms of insertion loss in dB across the frequency bands. The characteristics of the material influence significantly the performance. Simulation of the performance in a FEM software was conducted to evaluate the parameters that impact the insertion loss of the silencer. The results of the simulation study are presented in this paper.

2 Background

Details of a typical elbow silencer are shown in Figure 1. The main focus is to evaluate the transmission loss of the sound as the flows traverses the corner, treated with porous materials.

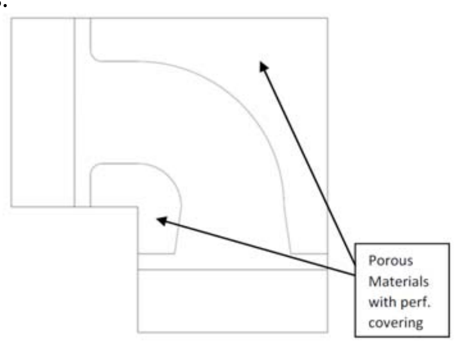

Figure 1: Schematic details of an elbow silencer.

The acoustic performance is evaluated by solving the governing wave equations between the inlet and outlet of the turning vanes. The porous materials are considered bulk materials and solution domain includes propagation within the bulk materials [1, 2, 3]. The insertion loss, IL, is given by,

$$IL = 10\log\left(\frac{L_{w,in}}{L_{w,out}}\right) \qquad [1]$$

where $L_{w,in}$ is the sound power at the inlet plane and $L_{w,out}$ is the sound power at the exit plane. FEM (Finite Element Analysis) methods were applied to evaluate the IL. The powerful software COMSOL Multiphysiscs was used as the FEM solver.

COMSOL Multiphysics is a powerful equation solver using FEM (Finite Element Methods) techniques. It results in very accurate results, provided the data for the input results are accurate. The only major disadvantage is the computing time and storage capacity of the machine used to solve the fundamental wave equations governing both the free air and porous material regions.

The propagation through the material requires the complex wave speed and acoustic density and they can be obtained from either of the two References 2 and 3. The main input parameter for the determination of the wave speed and density is the flow resistivity of the porous material.

To test the parametric analysis of elbow silencers, the experimental results of Itamoto et.al. will be used for comparison with COMSOL results [5]. The details of two different types of elbows and their inclusion as a single and double elbows are shown in Figures 2, 3 and 4 respectively.

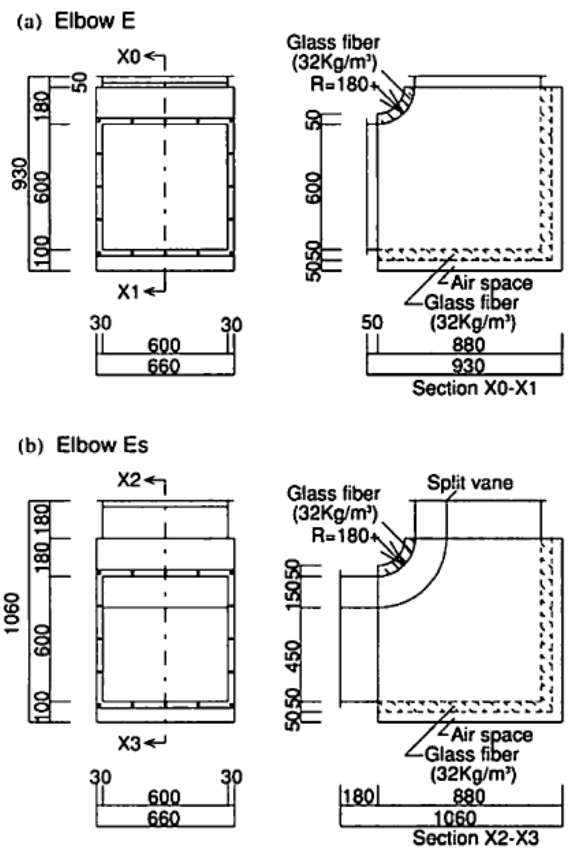

Figure 2: Details of elbow silencers (From Reference 5).

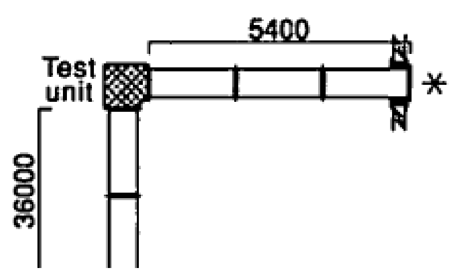

Figure 3: Single elbow (From Reference 5).

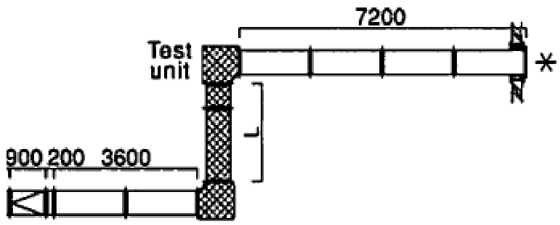

Figure 4: Double elbow (90º turn) (From Reference 5).

A number of the elbow combinations were modelled in COMSOL as 2-D and 3-D models and the results are discussed below.

3 Results and discussion

The results of the comparison are shown in Figure 5 and 6 for single and double elbows.

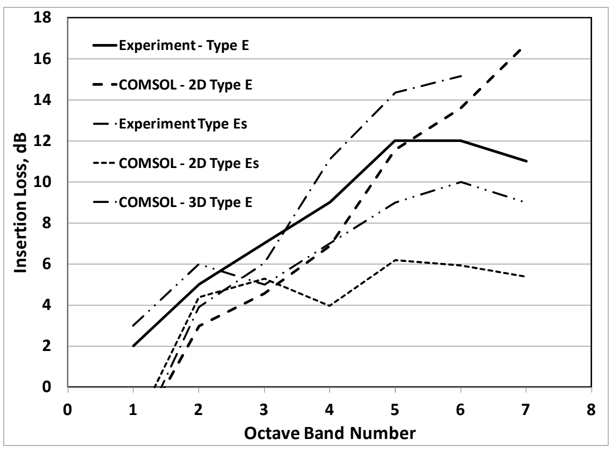

Figure 5: Comparison results for single elbows.

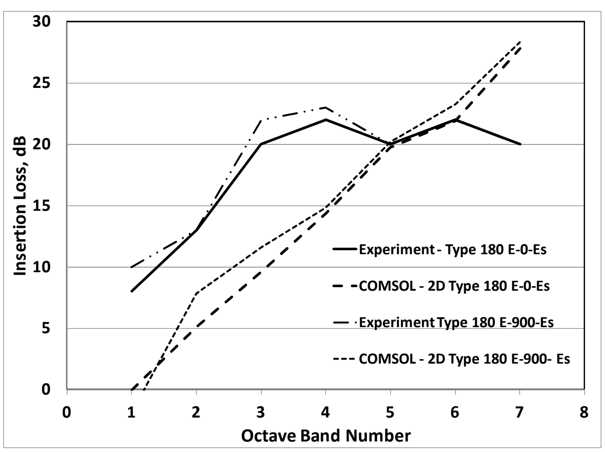

Figure 6: Comparison results for double elbows.

It is clear from Figures 5 and 6 that numerical method results do not compare very well with experimental data. It must be pointed out the flow resistivity of the porous material used in the experiments (Reference 5) was assumed to be 20,000 MKS rayls. The FEM models were applied again with different flow resistivity values. The results are highlighted in Table 1 below.

The results of Table 1 clearly show the importance of flow resistivity of the porous absorber. Further, the test results of Logawa and Hodgson showed that the flow resistivity is a function of frequency and can vary by about 10% to 15% across the frequency bands. Three dimensional modelling is also critical to provide results with engineering accuracy.

Condition	Octave band Number				
	1	2	3	4	5
180 deg E-900-E - Experiment	2	10	21	24	25
180 deg E-900-E - Prediction 3D (18,000 MKS Rayls)	0	7	17	31	28
180 deg E-900-E - Prediction 2D (18,000 MKS Rayls)	0	7	26	20	21
180 deg E-900-E - Prediction 3D (5,000 MKS Rayls)	0	6	17	26	26

Table 1: Insertion Loss of Double Elbow Silencers.

4 Conclusion

The insertion loss results of elbow silencers were determined from experiment and numerical analysis. The results showed the importance of flow resistivity of absorbers.

Acknowledgments

We would like to acknowledge that the work was conducted under a research grant from the Ontario Centre of Excellence with Kinetics Noise Control as the industry partner. We would like to thank Prof. Hodgson and his research group at the University of British Columbia for providing performance results of porous materials.

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SOUND PROPAGATION FROM WIND FARMS

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1 Introduction

There has been a significant increase in the installation of wind farms in Ontario. Wind farms will be a major energy supplier in Ontario due to the strong green-energy initiative of the provincial government. Noise impact of the wind turbines on nearby receptors has been a contentious issue. The current assessment procedures limit the methods to simple application of ISO-9613-Part 2. Detailed applications of terrain and weather information are beyond the scope of ISO-9613-part 2 procedures. Two improvements to the simple procedures are used to evaluate the receptor locations noise levels from a 5-turbine wind farm in southern Ontario. Comparisons between the three methods are presented in this paper.

2 Background

Simple procedures have been conventionally applied to evaluate the wind farm noise levels at nearby residential receptors [1, 2]. Many of the constraints assumed by the simple procedures are easily not complied with when such procedures are applied to wind farms. One constraint is the height of the wind turbine hub, usually in excess of 50 metres. Many of the receptors are located beyond the allowable distance of the simple procedures.

As part of the major research study of a 5-turbine wind farm in West Lincoln, Ontario, investigations were carried out to evaluate the noise levels at receptors from the wind farm. The results of the investigations are described below.

3 Noise Propagation Models

3.1 Cadna_A model

Cadna_A (Computer Aided Noise Abatement) is a leading software for calculation, presentation, assessment and prediction of environmental noise [3]. The procedures of ISO9613-Part II are applied in the evaluation. All the desired sources and receiver locations, the sound power level of the source are defined by the user. The weather data is defined in terms of the humidity, temperature, wind speed, wind direction, and temperature gradient, etc.

3.2 NORD 2000

The platform is designed for predicting noise level generated from a stationary source, infinite straight road or rail track sources [4]. The model is based on geometric ray theory and diffraction. The calculations are carried out in one-third octave bands. The sound power level of the source for various frequencies is defined along with the height of the source. Terrain profile is defined by the distance from the source, height, and ground type and roughness. Also, if there any scattering zone in the terrain, it can be categorized as forest area or housing area. The weather data is defined in terms of wind speed, wind direction, wind speed, turbulence strength, standard deviation of wind speed, temperature, temperature gradient, standard deviation of temperature gradient and turbulence strength (temperature).

3.3 Parabolic equation solver

The INPM noise model was developed by JASCO Applied Sciences, originally for use by the Canadian Department of National Defence as an environmental impact assessment and forecasting tool [5]. The INPM propagation modelling algorithm involves numerical computation of the parabolic form of the acoustic wave equation in vertical planes, thus providing a full description of the sound pressure in the air column along a radial from the source. The model takes fully into consideration the physical properties of the propagation media resulting in the complex structure of sound distribution.

INPM accounts for the acoustic influence of vertical profiles of atmospheric parameters (temperature, pressure, humidity) and is able to account for the influence of atmospheric turbulence. The model inputs the ground elevation and terrain type or cover (from which acoustic ground impedance is computed) from geo-referenced files to account for the local influence of these parameters on sound waves as they interact with the terrain. Sound propagation in each frequency band is computed independently and the results are summed to provide broadband estimates. Planar maps of noise footprints are obtained from multiple runs of the model along a fan of propagation radials from a source, a process that can be further expanded to multiple sources; the sound intensity is estimated at a standard receiver height (ear level) above ground.

4 The Wind Farm

The wind farm under investigation is located in the Town of West Lincoln, near Hamilton, Ontario. The layout in West Lincoln is shown in Figure 1. The wind farm and the five turbines became operational in May of this year. The distances from each turbine to each of the four receivers are shown in Table 1. SENES Consulatnts of Markham, Ontario evaluated the detailed weather data for three days for the month of June. Wind speed profiles evaluated by Senes Consultants at the four receiver locations are shown in Figure 2 for 8.30 at night.

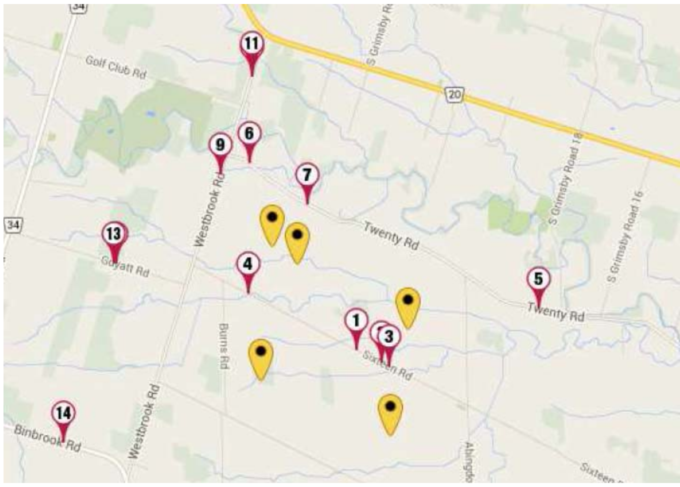

Figure 1: Wind turbines (arrows) and receptors.

	Receiver 1	Receiver 2	Receiver 3	Receiver 4
	ID - 1	ID - 11	ID - 5	ID - 14
WT1	1780	2080	3080	3190
WT2	1310	2420	2800	3010
WT3	620	3110	1440	4040
WT4	1600	3310	3450	2080
WT5	960	4440	2150	3710

Table 1: Distances between wind turbines and receptors.

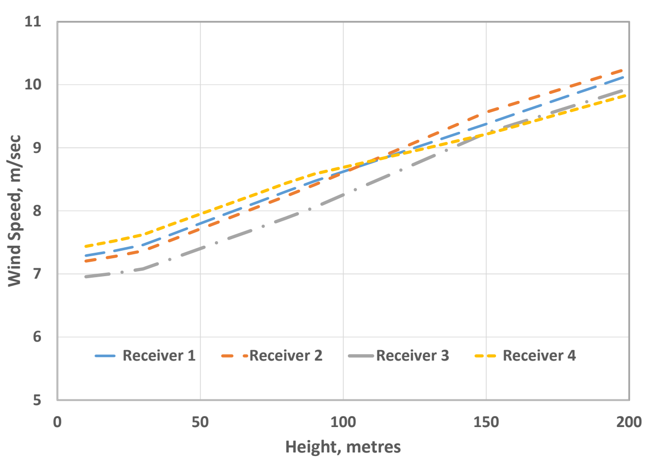

Figure 2: Wind Speed Profile at the four receptors.

The three propagation models were applied for a particular time in June and the sound levels were evaluated in dBA. The results of the evaluation are shown in Table 2 below.

Receptor	Cadna_A	NORD2000	INPM
R1	38.1	38.1	49.3
R2	26.3	23.1	35.7
R3	22.5	26.1	39.2
R4	20.1	21.9	35.3

Table 2: Predicted sound levels at receptors, dBA

5 Results and discussion

The main finding of the simulation is that results of CADNA_A and NORD2000 are seen to agree within engineering accuracy. However, INPM results are at least 10 dB higher. Unattended long-term monitoring was conducted at the four receptor locations after the turbines became operational in May of 2014. Preliminary results show that wind farm produced levels at the receptor that varied between 35 and 40 dBA. Measurements are still undergoing. In addition to long term measurements, spot measurements of noise levels as well as weather will be conducted.

Each of the prediction procedure are currently being refined. It will be difficult to determine the correct method of predicting results at the time of writing this paper.

6 Conclusion

The receptor sound levels at four residential locations were evaluated using three prediction models from a small wind farm of five wind turbines. The results show that two of the three methods predict lower noise levels.

Acknowledgments

We would like to acknowledge that the work was conducted under a research grant from the Niagara Region Health Unit, Ontario. We would like to thank Dr. Roberto Rocca and JASCO Applied Sciences for the use of the proprietary noise propagation code. We would like to thank Dr. Jim Young and SENES Consultants for providing detailed weather data.

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END REFLECTION LOSS IN REAL DUCT SYSTEMS

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1 Introduction

Lighter building construction has contributed to a focus on low frequency (LF) noise. The LF is difficult to predict within a noise path analysis and potentially more arduous to mitigate once a mechanical system is installed. For fans and air distribution systems, there is a palpable need for a better understanding of LF as a means to improve equipment performance, sound quality and occupant comfort. One of the key considerations in LF noise control is the amount of attenuation realized by the duct termination. The current prediction tools indicate that a significant amount of LF sound will reflect back through the duct when a sudden area change is encountered. Limited research is available on system interactions for this vital acoustic phenomena found in most ventilation systems. This study may help improve the assessment of LF duct reflection in real systems.

2 Background

The LF physical phenomena called End Reflection Loss (ERL; dB) may be described as the apparent change in sound power observed at a duct termination, as derived by the difference between the incident and reflected sound power at the duct termination or boundary. ERL occurs in the plane wave region, defined at frequencies below the cut-off (f_c; Hz) for ducts or openings:

$$f_c = \frac{c}{2De} \qquad (1)$$

where: c = speed of sound in air (m/s); and,
De = equivalent duct diameter (m).

One of the earliest descriptions of ERL was found in the book Noise Reduction [1]. This was followed by handbooks published by the American Society of Heating, Refrigeration, and Air-Conditioning Engineers (ASHRAE), called the Guide and Data Book [2]. This appears to be the basis of the ERL used today, including the current ASHRAE Handbook for both a flush termination and one that extends into a space. Recent research [3] has enhanced the understanding of ERL. This work included various termination conditions (diffusers, grilles and flexible ducts). This work did not focus on the impacts, if any, from the conditions further upstream from the termination. ERL is based on analytical assumptions for long, straight duct sections. ASHRAE provides a cautionary note: “Many air distribution systems and fan equipment do not have long straight sections (greater than 3xDe)…effects of these configurations on ERL are not known”. The literature does not provide any further guidance on the system related impacts. Real systems rarely provide more than 3 x De of uniform duct immediately ahead of the termination. For example, a fan intake may be open to atmosphere, and HVAC systems normally include elbow turns and off-sets between the fan and the termination. Uniform plane wave action may be rare in real systems, thus ERL may be subject to End Reflection System Effects (ERSE).

3 Experimental Study

This study was not configured to measure ERL directly. The effects could be observed indirectly through a controlled system configuration. The testing was conducted in the *VAW Systems Noise Control Applications Laboratory* (NCAL), located in Newmarket, ON. The apparatus is based on the applicable sections of the ISO 7235 [4] and the ASTM E477 [5] standards. The set-up (Figure 1) includes a large, absorptive source chamber housing loudspeakers that produce broadband pink noise. The source is connected to a modular wind tunnel consisting of heavy steel walls. The NCAL offers the ability to readily modify the wind tunnel into straight, elbow and multi-elbow configurations. Two duct configurations (Table 1) were assessed: (1) straight; and, (2) double-elbow. The uniform duct between the miter elbow and the termination was 8xDe. The measurements were made with a sound intensity system, including both in-duct and end-of-duct sound intensity levels (Li; dB). The sound power levels (Lw; dB ref10^{-12} watts) in this study are derived from the Li data as per the ISO 9614-3 [6] standard.

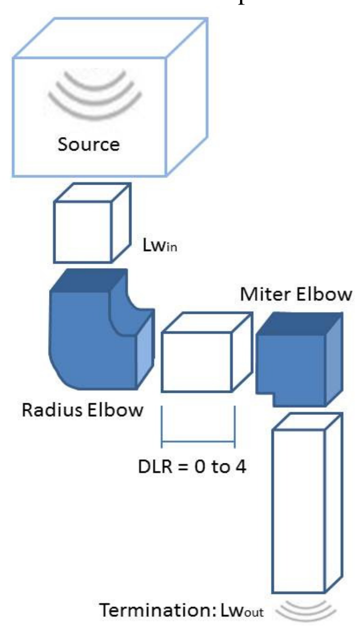

Figure 1: Experimental Set-up – Double-Elbow

Configuration	Size (mm)	Cut-off Freq. (Hz)
Straight	900x600	207
Double Elbow	600x600	253

Table 1: Wind Tunnel Set-up

The straight duct layout is rarely encountered in real systems, thus may be considered a baseline condition. The length of the wind tunnel was set to a Duct Length Ratio (DLR) of 0, 6, 10 and 16, where DLR may be defined as:

$$DLR = \frac{WindTunnel\ Length}{De} \qquad (2)$$

The double elbow layout allows for an evaluation of a common interaction between a radius and miter fitting. The space between the elbows was set to a DLR of 0, 1, 2 and 4.

4 Results

The measured Lw at the termination (Lw_{out}) is corrected (Lw_{out*}) at each 1/3-octave band frequency with the attenuation of the additional duct (IL_{DLR}), as follows:

$$Lw_{out*} = Lw_{out} + IL_{DLR} \qquad (3)$$

The straight wind tunnel resulted in a near constant sound power into the system (Lw_{in}) for changing DLR. As highlighted in Table 2, the standard deviation (SD, dB) of the Lw_{in} was significantly smaller than the corresponding SD of the Lw_{out*} for the measurements taken at four DLR. Similar results were found for the double elbow configuration. Over the range of DLR in this study, the Lw_{in} was relatively constant, while the corresponding Lw_{out*} varied significantly for relatively small system changes.

Measurement Location	Third-Octave Band Frequency (Hz)					
	50	63	80	100	125	160
Lw_{in}	0.69	0.33	0.10	0.33	0.17	0.18
Lw_{out*}	1.07	2.57	1.39	0.97	0.93	0.42

Table 2: SD (dB) of Lw_{in} and Lw_{out*} for Straight Duct

The Lw_{out*} may be investigated closely while under apparent constant Lw_{in}. As shown in Figure 2, the LF Lw_{out*} varied at a level close to the corresponding ERL.

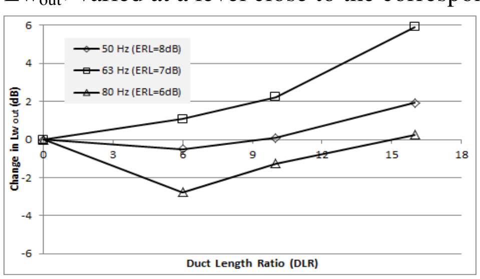

Figure 2: Straight Duct Lw_{out*} for increasing DLR (50,63&80Hz)

As shown in Figure 3, the double-elbow configuration resulted in a similar Lw_{out*} trend for a small change in DLR.

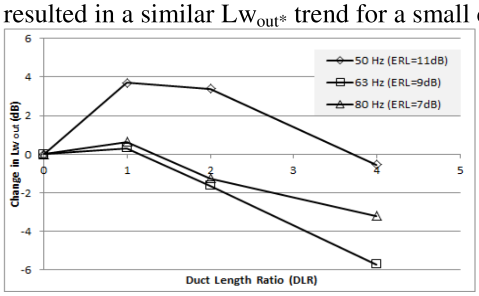

Figure 3: Double Elbow Lw_{out*} for increasing DLR (50,63&80Hz)

There is an apparent spread in the corrected sound power at the duct termination when the conditions upstream are varied. We may define the ERSE (Table 3) as the range in Lw_{out*}, as observed for various DLR, as follows:

$$ERSE = Lw_{out*}\ Max - Lw_{out*}\ Min \qquad (4)$$

The Li measurements were taken at various locations, consisting of a microphone traverse that divided the section into 9 equal areas. This process created Li contours and were evaluated as: “Very Uniform” (all 9 measurements within +/- 1 dB); “Uniform” (all 9 measurements within +/- 2 dB); and, “Not Uniform” (>3 dB spread). Changing the space between the two elbows form DLR = 0 to 2 did not impact the Li contours near the source and at the termination. The corresponding Li contour at the miter elbow changed from Very Uniform to Not Uniform at 50 and 63 Hz. Virtually all of the contours at > 63 Hz were found to be Very Uniform for all measurement locations.

Duct System Configuration	Third-Octave Band Frequency (Hz)					
	50	63	80	100	125	160
Straight	2.4	5.9	3.0	2.3	1.8	0.9
Double Elbow	4.3	6.0	3.8	2.6	0.9	2.0

Table 3: Range in Lw_{out*} (ERSE; dB) for Apparent Constant Lw_{in}

5 Conclusion

The impacts on ERL may be observed through system duct changes. The configurations noted in this study are relatively straightforward, but resemble some real-life systems. In a controlled laboratory set-up, conditions ahead of the termination appear to significantly impact the apparent ERL, resulting in ERSE. In-duct measurements may provide some value in terms of identifying conditions where the ERSE may emerge. Users of the analytical ERL tools should apply the values with caution, as the ERSE appears to be close to the magnitude of the ERL itself. Further research may look at a potential relationship between the sound field in the duct and the resulting ERSE.

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SUPPRESSION OF ACOUSTIC RESONANCE IN PIPING SYSTEM USING PASSIVE CONTROL DEVICES

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1 Introduction

Generation of pressure pulsations in piping system is often arising from the use of reciprocating pumps and/or the dynamic instability of valves, which leads to unwanted noise and vibration problems [1]. When the frequencies of these pulsations match the resonant frequencies of the piping system, this may result in an acoustic resonance which can produce large pressure fluctuations, often several times the dynamic head of the flow in the main pipe, as well as vibration of the piping system and surrounding components. A commonly used technique to reduce these pulsations in the piping system is the use of passive acoustic dampers such as Helmholtz resonators, and quarter wavelength side-branches. The design of these devices is well understood from a theoretical standpoint [2]. Usually, they are large and problematic to implement at certain locations. Thus, this study aims to investigate the effectiveness of different passive control devices, specifically Helmholtz resonators and quarter wave side-branches for attenuation of pressure pulsations in piping system.

2 Experimental Setup

The experiments were conducted using a 3.740 m long PVC pipe of 0.1016 m internal diameter, Figure 1**.** A speaker (acoustic excitation source) is attached at the end of the pipe to excite its acoustic modes. Pressure fluctuations at two positions are measured using flush mounted condenser microphones. A single or a pair of Helmholtz resonators and quarter wave side-branches can be attached at several positions of the tubes. For the current experiments, two different sizes of Helmholtz resonators are used, one with a frequency of 108 Hz (HR108) and another with a frequency of 178 Hz (HR178). Also side branches (QW108 and QW178) are designed for suppression at the same frequencies.

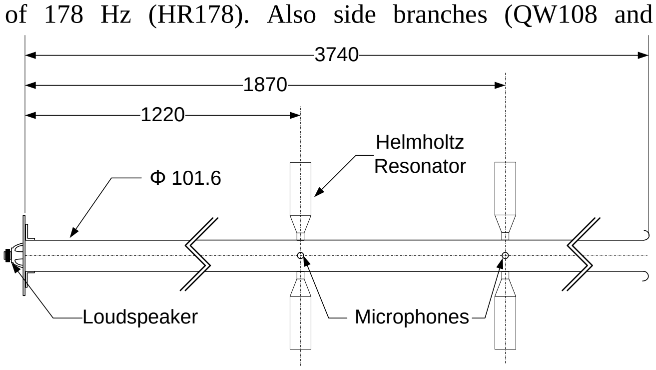

Figure 1 : Pipe parts assembled in one of the possible test configuration.

3 Results

3.1 Experimental results

The response of acoustic modes to attachment of several Helmholtz resonators is measured experimentally for several combinations of resonators at different positions, namely, at the middle of the pipe and at one-third of its length. The experiments show that placing the resonators at the middle of the pipe has a minor attenuation effect on the acoustic modes close to resonators frequencies. For example, with a single HR178 placed at the middle of the pipe, the 4th and 5th acoustic modes pressures, located at 168 Hz and 214 Hz are slightly changed with 16% increase and 10% decrease; respectively, compared to the base case where no Helmholtz resonators are attached. The same trend is also noticed with placing a HR108 at the middle of the pipe. On the other hand, when placing the resonators at one-third pipe length location far from the speaker, a more pronounced effect is obtained as shown in

Figure 2 for different combination of resonators. The attachment of several multiple Helmholtz resonators of the same size enhances the effect obtained from attaching a single resonator. This effect is found not to be a simple superposition as reported by Biswas and Agrawal [3] but it was found to follow a trend similar to a diminishing marginal utility. Also, the effect of attaching two different resonators is found to be a combination of the effect from attaching each of them alone.

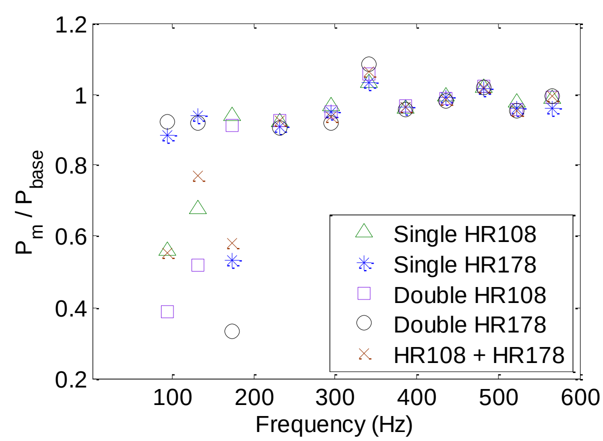

Figure 2: Normalized pressure response of acoustic modes with Helmholtz resonator(s) attached at one third the tube length from speaker. (2nd mode: 90 Hz, 3rd mode: 132 Hz, 4th mode: 168 Hz)

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3.2 Numerical Simulation

To investigate the effect of changing the resonators location along the piping system on its acoustic response, numerical simulations for the apparatus response by DELTAEC are performed for the system without and with attaching Helmholtz resonators and quarter wave side-branches. Figure 3 shows that the closest acoustic modes, 4^{th} and 5^{th} modes, are highly attenuated when the side branch and Helmholtz resonators are attached at the acoustic pressure antinodes where x/λ ratio is a multiples of 0.5, and can cause an adverse effect if each of them is attached at the pressure nodes. However, the effect of attaching a side branch on the system response is stronger than a Helmholtz resonator for both suppression and enhancement of acoustic pressure. This behavior agrees with the normalized pressure response obtained from the experimental results at the middle and the one-third pipe locations. The same conclusion can be drawn from Figure 4, which represents the effect of the location that these control devices have on modes shifting.

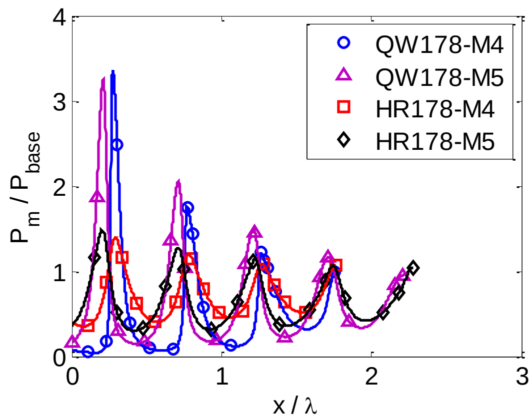

Figure 3: The effect of normalized position of the HR178 resonator and the QW178 quarter wave side-branch on the normalized pressure amplitude relative to no attachment case for the 4^{th} mode (M4: 168 Hz) and 5^{th} mode (M5: 212 Hz).

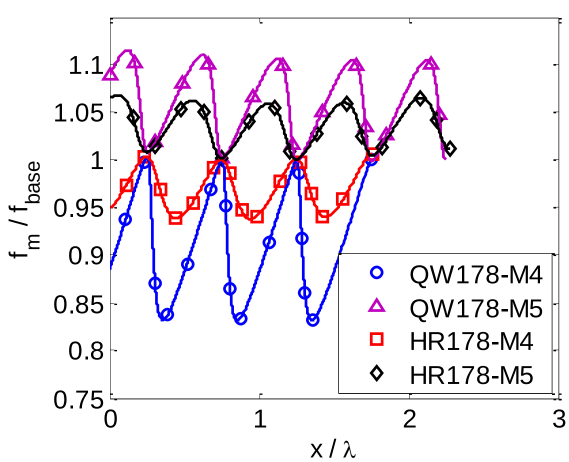

Figure 4: The effect of normalized position of the HR178 resonator and the QW178 quarter wave side-branch on the normalized frequency relative to no attachment case for the 4^{th} mode (M4: 168 Hz) and 5^{th} mode (M5: 212 Hz).

It is noticed that the attachment of these control devices shifts the frequencies of the modes *away* from the resonance frequency of the passive control devices themselves, and the closest modes to the device frequency are the most to experience shifting. The same analysis was performed for attaching HR108 and QW108, and it shows the same behavior for all modes' pressure amplitudes and frequency shifting.

4 Discussion

It has been noticed from an analysis done similar to what is shown in Figure 3 for higher modes that if the resonator or side branch is placed at a position where x/λ > 2.125 from the excitation source, a slight increase in the mode's acoustic pressure occurs. This is also proven from the experiments as shown in Figure 2. Attenuation for modes higher than 590 Hz is found to be less than 5% when the resonators are attached at one-third tube length location, this corresponds to x/λ = 2.115. Moreover, for the case of attaching two resonators of the same size, the overall change, reduction or increase, of normalized pressure along the various acoustic modes can be related to the change done by a single resonator from the following empirical relation obtained from Figure 2:

$$\Delta P\%|_{two\ HR} = \Delta P\%|_{single\ HR} \times (1 + 0.416) \qquad (1)$$

5 Conclusion

The effect of attaching several Helmholtz resonators and quarter wave side-branches, as acoustic passive control devices, is investigated numerically for both and experimentally for the Helmholtz resonators. The study found that quarter wave side branches have higher ability to suppress targeted acoustical modes than Helmholtz resonators, and the amount of suppression is highly dependent on devices locations relative to the acoustic pressure nodes and antinodes, which can cause an adverse effect if placed near the pressure nodes.

Acknowledgments

The authors acknowledge the financial support provided by the Natural Sciences and Engineering Research Council of Canada (NSERC).

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LOW SOUND LEVELS MEASURED IN REMOTE AREAS

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1 Introduction

Rural noise guidelines in Canada generally set a lower limit based on practicality and activity disruption rather than audibility. For example, the lowest sound level limit in Ontario[i] and Quebec[ii] is 40 dBA $L_{eq,night}$. Alberta[iii] uses 40 dBA, down to 30 dBA in quiet areas, and BC's guideline[iv] is virtually a copy of Alberta's. Health Canada[vvi] generally sets a lower limit of 45 dBA LDN, which would correspond to a limit of 37 dBA for a steady industrial sound (although they also recommend an additional 10 dB penalty in very quiet areas). Other provinces usually base any assessments on one of these limits. These lower limits are reasonable in not affecting activities such as speech or sleep however they should not be assumed to prevent the sound from being audible. If these values were much quieter they would in any case effectively prohibit industry from operating in rural areas. For example, a single front end loader with a sound power level of 110 dBA would produce about 37 dBA at 1 km. It is hard to see how mines, farms, drilling, etc. could reasonably occur with much lower limits in rural areas unless there were no inhabitants nearby, in which case noise limits would not be necessary in any case, except perhaps for wildlife. Complainants in rural areas, especially city dwellers seeking to escape the city often expect that no man-made sound be audible (except of course their own). If we simply define this as the intrusive sound being less than the quietest background we can see what this means in practice.

Many references assume rural sound levels in the 25 to 35 dBA range. The following is a typical example:

Table 1 - Misleading Sound Level Table

Outdoor Location	Average Outdoor Sound Levels (dB)
Apartment next to freeway	88
1 km from touchdown at major airport	82
Downtown with some construction activity	79
Urban high-density apartment	77
Urban row housing on major avenue	68
Old urban residential area	59
Wooded residential	51
Agricultural cropland	44
Rural residential	39
Wilderness ambient	35

Source: Protective Noise Levels: Condensed Version of EPA Levels Document EPA 550/9-79-100

One common source of error in these types of table appears to be their assumption that the numbers quoted are typical sound levels, while they appear to be based on the EPA Levels Document which clearly states that the values were measured as L_{DN}, which means that during the night time, which is generally the quietest time of day, the equivalent sound levels were increased by 10 dB. The measurements were also made in the early 70's, which may have meant the instrumentation noise floor was a consideration, although the B&K 2009 sound level meter of that time was capable of measuring 15 dBA with a one inch microphone and the author did so on a dock in Muskoka on a calm evening, even though a pump was clearly audible at the time about 500m across the lake. Similar results have also been seen in winter in the country when there is no wind and may actually have gone lower.

To sort out the actual sound levels in quiet rural areas we have made a number of measurements. Several examples are given below.

The first was a site several hundred meters inland from a marina at the end of a rural road in Northern Manitoba. The boats are generally not used at night and the site is more than 10 km from the nearest town and likely from any other man-made sound. Table 1 shows the lowest results measured in one night. The lowest $L_{A90,5min}$ measured was 17dBA and the lowest $L_{Aeq,5min}$ was 18 dBA. However the measured noise floor of this instrument was 15.8 dBA and when this is subracted logarithmically from the measurements we get 11 dBA and 14 dBA respectively.

Bear in mind that the 11 dBA really means somewhere below the noise floor of the instrument.

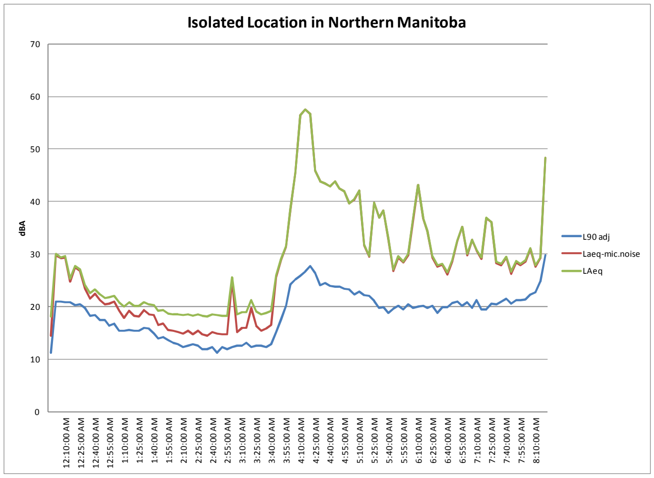

Figure 1 Isolated Location in Northern Manitoba

The second location was monitored for 4 nights in a trailer park in cottage country outside the City of Sudbury, about 3-5 km from any highways during the week when many of the residents were absent. This is a small vacation community. The quietest $L_{eq,10min}$ was 24 dBA and the quietest $L_{90,10min}$ was 22 dBA. Subtracting the instrument noise floor takes 1 dB more off each.

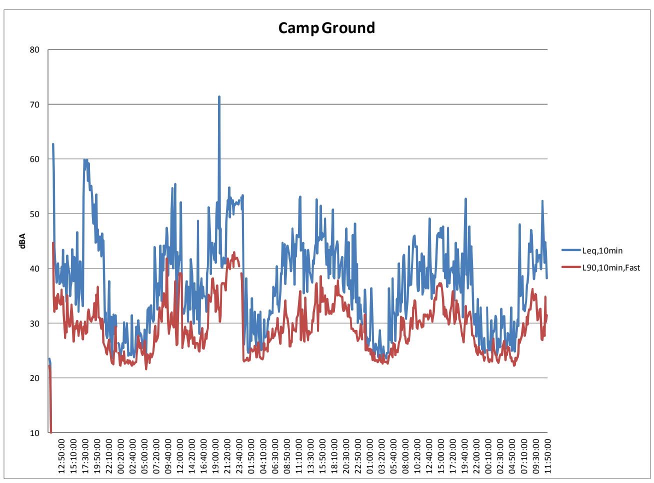

Figure 2 Campground Outside Sudbury

2 Conclusions

It is clear that sound levels can fall very low in rural and wilderness areas. The background noise in urban areas makes a suitable limit against which to judge intrusive noise and most urban and suburban noise limits either explicitly or implicitly end up with values approximating this background. In rural areas this is not possible if any development is to take place, except in very isolated locations. However residents, especially in cottage country and rural retreats often do judge intrusive rural noise by their ability to hear it and in some wilderness parks this may be a possible limit but it must be recognised that it cannot be used in many rural instances. What is important is that proponents, consultants and regulators do not compound this by implying that their limits do approximate the background in these areas.

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ii Quebec - Traitement des plaintes sur le bruit et exigences aux entreprises qui le génèrent - Références légales : LRQ (c. Q-2), articles 20 et 22, JUIN 2006

iii Alberta - AUC Rule 012, AER 038

iv British Columbia - Noise Control Best Practices Guideline

v Health Canada - Useful Information for Environmental Assessments – Noise Effects

[vi] Health Canada - Guidance for Evaluating Human Health Impacts in Environmental Assessment: Noise – DRAFT, January 2011

Exploring The Impacts Of Consistency In Sound Masking

Niklas Moeller

Electronic sound masking systems control the noise side of the signal-to-noise ratio in interior environments. Their effectiveness relates directly to how consistently the specified masking curve is achieved. Current system specifications generally allow a relatively wide range in performance, in large part reflecting expectations set by legacy technologies. This session presents a case study of sound masking measurements and speech intelligibility calculations conducted in office spaces. These are used as a foundation to discuss the impacts of local inconsistencies in the masking sound and to begin a discussion of appropriate performance requirements for masking systems.

Innovative Ways To Make Cross Laminated Timber Panels Sound-Absorptive

Banda Logawa, Murray Hodgson

Cross Laminated Timber (CLT) panels typically consist of several glued layers of wooden boards with orthogonally alternating directions. This cross-laminating process allows CLT panels to be used as load-bearing plate elements similar to concrete slabs. However, they are very sound-reflective, which can lead to concerns about acoustics. This project is aimed at investigating ways to improve the sound-absorption characteristics of the panels by integrating arrays of Helmholtz-resonator (HR) absorbers into the panels and establishing design guidelines for CLT-HR absorber panels for various room-acoustical applications. To design the new prototype panels, efforts have been made to measure and analyze the sound-absorption characteristics of exposed CLT surfaces in multiple buildings in British Columbia, investigate suitable methods and locations to measure both normal and random incidence sound-absorption characteristics, study the current manufacturing method of CLT panels, create acoustic models of CLT-HR absorber panels with various shapes and dimensions, and evaluate the sound absorption performance of prototype panels. This paper will report progress on this work.

HEARING CONSERVATION - CONSERVATION DE L'OUÏE

COMFORT FROM HEARING PROTECTORS

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1 Introduction

The two properties of utmost importance when dealing with hearing protectors are sound attenuation and comfort. Sound attenuation, defined as the difference between the sound levels of the open and the protected ear is well defined and there are standardized methods for its measurement.

There is no clear definition, however, on hearing protector comfort, nor are there standards for its measurement. Also, research done regarding comfort appears to be limited. A search conducted on the popular Web of Science database, shows that between the years 1970 and 2014, 208 papers were published dealing with attenuation, while there were only 22 papers related to comfort.

Comfort is an evaluation that may be defined quite simply as "all is well." It is inherently subjective in nature. It is also dependent on issues other than the protector itself, but related to the environment, such as temperature and humidity of the workplace. Other factors to be considered include the need for speech intelligibility and anatomical differences among wearers.

In contrast, discomfort may be defined (and measured) along many dimensions. Some examples of those dimensions are shown in Table 1.:

Pressure on the eardrum
Irritation of the canal lining
Feeling of fullness
Occlusion effect
Heaviness
Heat and humidity
Pressure against the head
Interference with headgear, eyeglasses and hair

Table 1: Some factors for discomfort

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Over the years researchers have taken different approaches to study comfort. Some did literature searches; other did lab and/or field studies. In general, the lab studies involved short exposure times. Nonetheless, some researchers claim that short exposures lead to similar outcomes than would be expected from longer exposures typical of the work environment.

When studying comfort of earmuffs, some focused on the force exerted against the head, while others were interested in pressure measurement. Also there are studies regarding the influence of heat and humidity on comfort. Finally, some researchers have tried to put together sound attenuation and comfort. Almost all of them made use of some sort of questionnaire to quantify the weight of the different factors of discomfort. Table 2 shows an example of a questionnaire.

How does the protector feel?

Painless	____	____	____	____	____	Painful
Tight	____	____	____	____	____	Loose
Heavy	____	____	____	____	____	Light

Table 2: Example of Bipolar Scales

This presentation critically evaluates studies on comfort in hearing protectors completed over the past 25 years, with the intention of developing a novel procedure for ranking hearing protectors on the basis of comfort under specified conditions. This approach will involve statistical regression, allowing us to evaluate the relative contribution of different factors on comfort.

2 Discomfort factors

As defined in Wikipedia, "*comfort (or comfortability, or being comfortable) is a sense of physical or psychological ease, often characterized as a lack of hardship". Further, "because of the personal nature of positive associations, psychological comfort is highly subjective*".

The definition of comfort involves several contributing factors that are not simple to define. For example, if we qualify a pair of shoes as "comfortable", there is no need to explain why: it is implicit that those shoes fulfill every requirement for feeling comfortable. However, if the shoes are not comfortable, then one must define the quality or the qualities that make them feel as such and those can be numerous. The same concept can be applied to hearing protectors. As shown further below, there are many factors that can make them uncomfortable.

As a start, we shall propose that no Personal Protective Equipment (PPE) is comfortable. All PPEs imply a certain degree of discomfort, probably related to the fact that they are “extra garments”, not needed for performing the task at hand.

There are several qualities that can cause discomfort. For instance, safety shoes are heavy and cumbersome. Hard hats are also heavy and particularly cumbersome to wear in a hot and humid work environment. Gloves can be uncomfortable in a similar environment. On top of perspiration, they may cause skin reactions such as eczema. Safety glasses can be considered the best accepted PPE. They are typically light and, because most people are used to wearing glasses, there is a little perceived discomfort or rejection

With respect to comfort, respirators and hearing protectors are often considered the worst offenders. Depending on the type, respirators tend to restrict breathing. Also they are often heavy and hot. However, respirators are perceived as life-savers. Very little effort is needed to convince workers that they must wear them for their own good.

The situation is different with hearing protectors: in general, as mentioned above, there is much resistance to their wear. A greater amount of effort and time is also required to build awareness regarding noise as a hazard. (« Ears don’t bleed »). In addition, HPDs often interfere with speech perception, warning sounds and alarms.

It should be noted that some comfort factors are a function of the length of time HPDs are worn. There are instances when devices appear comfortable when they are first donned. After a while, however, the user may start feeling the weight/pressure and finds them burdensome. This is often the case with muffs, particularly when cap-mounted. Other types of hearing protectors, on the contrary, feel uncomfortable when they are first donned, but after a while, the wearer forgets that he has them on. This is often the case with earplugs.

Based on the above considerations, one would expect that comfort studies should be performed over extended periods of time.

3 Conclusions

The issue of comfort (or discomfort) is complex and appears to be governed by subjective and objective factors:

a) Subjective.
   Different subjects perceive and react differently to the same protector.
b) Objective.
   a. Anatomical differences
   b. Duration of use
   c. Environmental factors (mainly heat and humidity)
   d. Types of protectors (plugs, muffs and semi-inserts)

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Did you say ”bionic” ear?

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Introduction

For decades, Hearing Protection Devices (HPDs) have been used to protect wearers against noise-induced hearing loss (NIHL). HPDs are available under various shapes and forms, from roll-down foam earplugs to cap-mounted earmuffs (see Fig. 1 for illustrations of common hearing protectors and their common classification). Until recently, HPDs have acted simply as passive acoustical barriers intended to prevent sound from reaching the ear canal. With the increasing miniaturization of electronic components and consolidation of consumer electronic goods, new electronic HPDs have come onto the marketplace. Thus, entirely new communication devices are being developed such as Hearing Aids that are also a wireless cellphone earpiece. The convergence of hearing protection devices, hearing aids and communication earpieces appears to be the next step and this technology is sometimes referred to as a “bionic” ear.

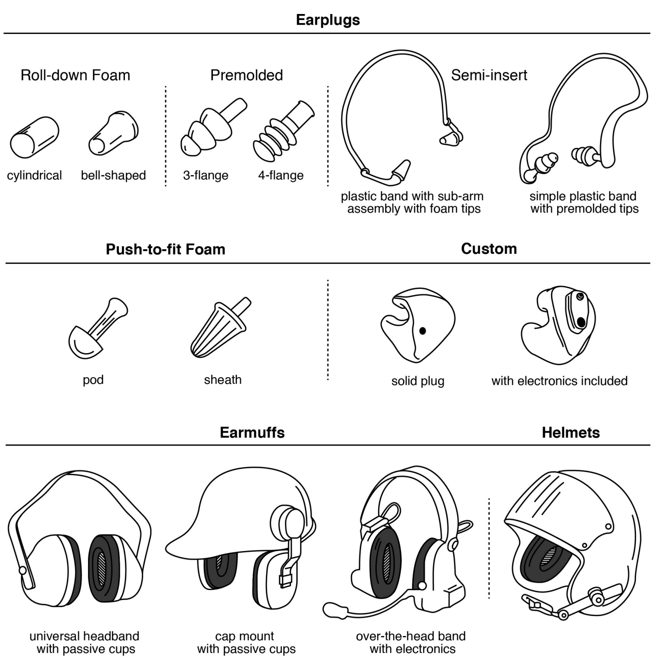

Figure 1: Illustrations of typical hearing protectors and their common classification.

1 Combining hearing protection, hearing aid and communication into one digital device

HPDs reduce the sound energy reaching the wearer’s eardrum and -in their passive linear form- cannot distinguish between noise and useful signals, such as speech and warning signals. This issue can now be addressed using electronic HPDs that use an external microphone a -preferably- digital signal processor and an internal loudspeaker. The electronic HPD could be an earmuff-type, while it could also be a custom earplug, such as the one illustrated in Figure 1 middle-right. Figure 2 illustrates the electro-acoustical components and equivalent schematic of a digital version of a custom electronic earplug.

1.1 Digital Hearing Protection

Many electronic HPDs have already been developed for the industrial and military workplaces, but only a few feature digital signal processing. Some of our current research efforts are oriented towards the detection of speech [1] and warning signals [2] in noisy environments, the denoising and enhancement of the speech signals [3], as well as the protection of musicians [4] [5] [6] and other special wearers [1].

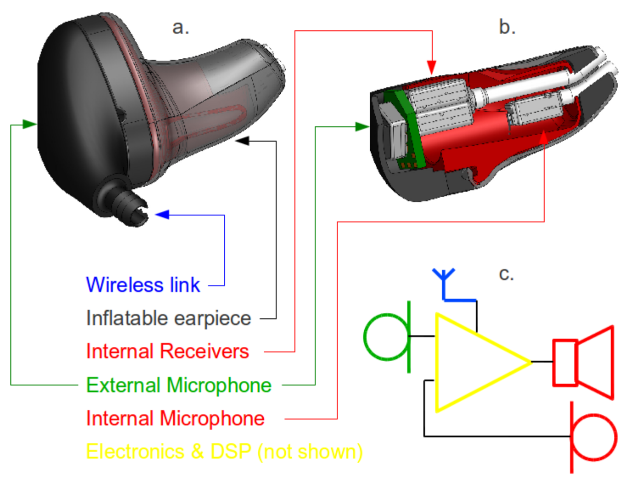

Figure 2: Overview of a digital custom earpiece (a), its electro-acoustical components (b), and equivalent schematic (c).

1.2 Digital Hearing Aid

Hearing Aids are considered in most jurisdictions as medical devices and usually require hearing specialists to dispense them. Nevertheless, the sound amplification they provide is very beneficial for speech perception and relatively easy to use in low-noise environments, giving use to a wide range of consumer products, sometimes referred to as Personal Sound Amplification Products (PASPs), ranging from by integrating revival of so-called *body-aids* with the use of modern smart phones to digital in-ear devices featuring a pre-programmed hearing aid. It should be noted that the use of hearing-aids or combined use of hearing aid and HPD in noisy environments are still challenging topics for hearing conservationists [7] and hearing-impaired workers in noisy environments would definitively benefit from a digital in-ear device combining hearing protection and hearing aid.

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1.3 Digital Communication Device

With the widespread adoption of cell phones and smart phones, earphones and communication earpieces are now ubiquitous in their wired and wireless versions. Despite intense research efforts in speech noise reduction algorithms and in multi-/noise-canceling speech capture microphones, communication in noise is often less than satisfactory. Many consumers find it difficult to hold a phone conversation on a busy street. Business travelers cannot comfortably join teleconference while in the ambient noise of an airport and it is still nearly impossible for many workers in noisy industrial environments to communicate safely over their personal radios without overexposing their hearing either to ambient noise or to a communication signal. To address these communication in a noise issues, the superior sound isolation offered by a custom-fitted in-ear device, illustrated in Figure 2, enables quieter playback, reducing the chances of overexposing the wearer's hearing [8]. Also, the in-ear microphone, illustrated in Fig. 2, could be used to capture the wearer's speech under the HPD with a favorable signal-to-noise ratio, and further spectral expansion techniques could be used to enhance the quality and intelligibility of the picked-up speech [9]. Finally, a Radio-Acoustical Virtual Environment (RAVE) could be created in civilian and military environments, where teams equipped with such digital communication in-ear devices could speak to each other on personal radios, while the speech signal of the wearer would only be broadcasted to receivers within a given radius, just like in normal open-air acoustic communication [10].

2 The earcanal : a nice place to be for ”wearable” technologies

While integrating all the above-mentioned audio applications could already open a new realm of technologies and products for consumers, medical and military applications, the earcanal is still probably the ideal location -currently underestimated- for many more technologies in years to come. The growing field of ”wearable technologies” aims at developing electronic devices that -broadly defined- interconnect the human body to the machine. Proof-of-concepts have been made to collect acoustic bio-signals -such as the heart beat and breathing rate- within the earcanal, for health and fitness monitoring applications. Miniature electrodes have recently been embedded on in-ear devices to enable portable and non-invasive electroencephalography (EEG) monitoring of brain-wave and other evoked responses. One could possibly foresee that a truly light and non-invasive Brain-Computer-Interface (BCI) could be developed by integrating acoustical and electro-physiological sensors. Interestingly, the earcanal may be one of the best places, to harvest energy to power -at least partially- all future ”bionic ear” and other ”wearable” devices, as recently demonstrated in a study on harvesting energy from earcanal deformation by jaw-joint movement [11].

Acknowledgments

The author is thankful for the financial support of the Sonomax-ÉTS Industrial Research Chair in In-Ear Technologies and for the relentless efforts of his collaborators and graduate students in this quest for a ”bionic ear”.

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EXPERIMENTAL STUDY ON CUSTOM EARPLUG RETENTION IN REAL-WORLD SITUATIONS

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1 Introduction

Every day, millions of workers wear Hearing Protection Devices (HPD) to protect themselves against noise induced hearing loss. For many reasons -listed in [1]- the attenuation provided by HPDs are highly variable across wearers and only individual fit-testing can measure the amount of attenuation achieved by a given HPD for a given individual. However, this measurement is in truth only a "snapshot" revealing the attenuation of a given HPD at the time of measurement. Nothing guarantees that this attenuation value will later be achieved by the wearer in the field. Studies have indeed demonstrated that earplugs are not always consistently fitted [2] and many hearing conservationists suspect that most earplugs work loose with time and require periodic resetting.

This paper addresses the issue of HPD retention, i.e. the HPD's ability to maintain its acoustic seal and attenuation over time, and presents the results of an experimental study of earplug retention in real-world situations. The earplugs chosen for these tests are custom molded earplugs, as these types of HPDs are highly personalized to fit the wearer's earcanal geometry and are often assumed to maintain their attenuation performance for longer periods than non custom earplugs.

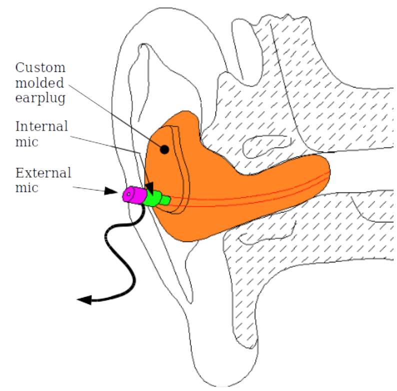

Figure 1: Overview of the custom earplug instrumented with the F-MIRE attenuation measurement device

2 Material and Method

2.1 Material Used

The earplug used in this study is a SonoCustom™V3 earplug, manufactured by Sonomax (Montreal, QC). The field attenuation estimation system (FAES) used during this experiment to fit-test the custom earplugs is SonoPass™, illustrated in Fig. 1. It performs a so-called field-Microphone-in-real-ear (F-MIRE) attenuation measurement, as a miniature microphone is temporary inserted in a sound-bore within the generic rigid core to measure sound pressure levels in the residual earcanal portion between the HPD and the eardrum. Attached to the back of this internal pressure microphone is an external pressure microphone so that sound pressure level difference across the earplug can be measured in the presence of loud pink noise generated from an outside reference sound source, frontal incidence. The attenuation measurement is performed by a trained technician after the end-user removes and replaces the custom earplug in order to perform the initial attenuation test, at time t=0.

2.2 Experimental Protocol

To assess earplug retention, the individual attenuation will be measured over time with the FAES system previously described. Between the initial attenuation test, at time t=0, and any subsequent testing, great care will be taken to not touch or purposely alter the placement of the earplug. The test-subject is instructed to go back to his/her daily activities, but to not touch or reposition the earplug. Similarly, the F-MIRE microphone doublet is left in place, i.e. inserted inside the earplug's sound bore, with its electrical wires carefully wrapped and secured on the test-subject's shoulder, so that the technician can easily retest the individual earplug attenuation at periodic intervals.

Two studies were conducted at Sonomax Hearing Healthcare laboratories in 2005, with two groups of in-house employees, as detailed in Table 1. While the first study was conducted on a rather long time span, the second was later conducted on a reduced time span, but with smaller periodic testing intervals, to better reveal the short-term retention issues.

Date	Nb Subj	Nb Ears	Step	Duration
May 2005	3	2	1 h	3 h
October 2005	4	1	20 min	2 h

Table 1: Details on the two studies conducted in 2005

3 Experimental Results

Fig. 2 represents the individual attenuation at each octave-band center frequency and overall (PAR) over time for the four subjects of the October 2005 study.

In order to assess the retention over time for the different subjects, the variation in attenuation values relative to the initial attenuation value, at t=0, will be computed in all subsequent figures. Fig. 3 presents the mean and standard de-

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viation of this attenuation variation for the May 2005 study, while Fig. 4 presents it for the October 2005 study.

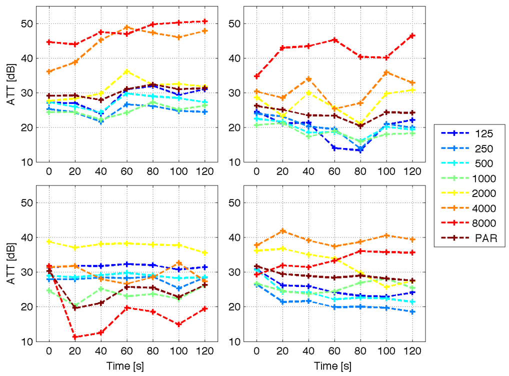

Figure 2: October 2005 - Individual attenuation at each octave-band center frequency and overall (PAR) over time

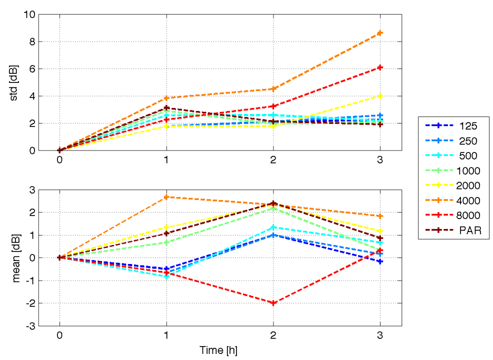

Figure 3: May 2005 - Mean and standard deviation of the attenuation variation at each octave-band center frequency and overall (PAR) over time

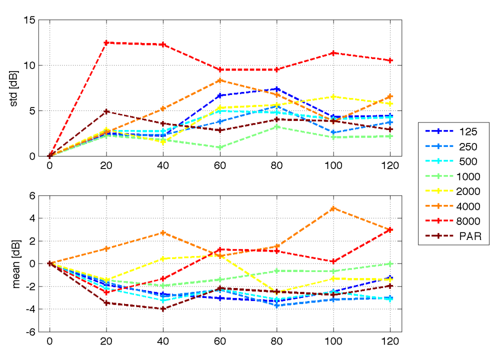

Figure 4: October 2005 - Mean and standard deviation of the attenuation variation at each octave-band center frequency and overall (PAR) over time

4 Discussions

For the first study (May 2005), the group mean attenuation variation remains smaller than 3 dB, over the 3 hour test period, but can be seen on all octave-band frequencies. The standard-deviation after 1 hour already reaches a 3 dB value for most frequencies and becomes even greater after 3 hours. It is particularly interesting to note that the 4 and 8 kHz octave-band are the frequency bands that vary most, as they contain the occluded ear canal resonances. The residual earcanal length might have slightly evolved over time, indicating a light displacement of the earplug within the earcanal.

For the second study (October 2005), the much smaller periodic testing interval shows that the group average attenuation varies very quickly, and reaches maximums in less than 40 minutes. What is of particular interest is that the group tends to lose attenuation over time, and the low frequencies, 125 – 500 Hz, are most affected by the -assumed- earplug displacement. The higher frequencies, 4 – 8 kHz, appear to behave differently, as their mean attenuation does actually increase over time to reach a maximum after 2 hours. The standard-deviations show that high frequencies are the most variable attenuation values, which could be again explained by the shifted occluded earcanal resonances.

5 Conclusions

This experimental study presented the findings on custom earplug retention in real-world situations and showed that custom earplugs may work loose very quickly after having been fitted (<40 min.). As custom molded hearing protectors are often assumed to maintain their attenuation performance over longer periods than non custom earplugs, this retention issue might even be more significant with non-custom earplugs. Further research is therefore urgently needed to collect such experimental data on non-custom earplugs. This data could guide the hearing conservationist to substantiate his recommendations for periodic resetting of earplugs. Both retention variability data on custom and non-custom earplugs could also be used and accounted for within fit-testing systems, so that the reported attenuation may be factored in regarding a correction for this field discrepancy.

Acknowledgments

The authors would like to thank Mr. Michael C. Turcot for releasing the experimental dataset initially collected at Sonomax Hearing Healthcare facilities in 2005. The first author is also thankful to the *Sonomax-ÉTS Industrial Research Chair in In-Ear Technology* for its financial support of the present work and re-analysis.

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A NEW CSA HEARING PROTECTION STANDARD

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1 Introduction

The first CSA hearing protectors' standard was published back in 1965. Since then, there have been 6 editions, the last being issued in 2002. Right now, a new edition has just gone through a mandatory Public Review as "*Z94.2-14 Hearing Protection Devices – Performance, selection, care and use*". It is presently being prepared for final review and approval, appearing probably before the year-end. The reason for that many updates has been the fact that the industry has greatly improved their product. There have been further advances in the way protectors are tested and the way test results are used for calculating the noise level of the protected ear. The biggest problem has been (and is still) how to equate the lab testing results with real life situations, where the attenuation is always lower.

While preparing the present edition, the writing Subcommittee has taken interest in certification of hearing protection devices (HPDs). Although the final document doesn't call for certifying the devices, the language has been prepared so that certification could be introduced in a later edition without major changes in the document.

This edition expands on performance requirements and rating schemes to help the user select the device most appropriate for a given work situation. It now includes a de-rating scheme for the widely used Noise Reduction Rating (NRR) designed to obtain more reliable estimates of the noise level of the protected ear. The potential use of field attenuation estimation systems (FAES) is also introduced.

2 Main Issues

Some of the outstanding issues in this standard are:

2.1 Attenuation Testing

Two testing methods are included in the standard. The first one is the Experimenter-fit real-ear attenuation, ANSI S3.19 -1974. It is needed for the calculation of NRR and for determining the Class of a protector.

The second is the Subject-fit real-ear attenuation, ANSI/ASA S12.6–2008, Method B (inexperienced subjects). Results from tests following this method are used for the calculation of SNR(SF_{84}), a rating proven to be closer to what is found in real life situations.

2.2 Protectors' classification

A) Class

A Class is assigned to a protector using results from ANSI S3.19 test results. This classification is kept mainly because it is included in some provincial legislation and also used by manufacturers and users alike. The Standard recommends the use of different classes for different ranges of noise exposure levels, making easy the choice of a class.

B) NRR

The most popular classification has been and still is the NRR (Noise Reduction Rating). The two main reasons are: a) the requirement by the USA EPA to have it written on the protector's package and b) the ease of calculating the noise level of the protected ear. It has been criticized because of the overly optimistic value, due mainly to the way the testing is done (using the ANSI S3.19-1974 method). Because the field attenuation is smaller than the lab results, the standard includes one scheme for de-rating of NRR, including several examples of its application.

C) SNR(SF84), Single Number Rating (Subject Fit 84th Percentile)

This is the same rating included in the previous issue of the standard. It is calculated using results from test done following ANSI S12.6 standard Method B that uses naive subjects. It has been demonstrated that sound levels of the protected ear calculated using that particular rating, approximate closely to those obtained in real life situations.

D) Octave-Band Computation

This computation was also included in the previous standard. It is the most complex of the four methods described above. It requires the ambient noise level to be measured in 1/1 octave bands, something easily done with the modern instrumentation. This method provides the greatest potential accuracy. It is recommended to be used every time the 8-hr equivalent noise exposure level exceeds 105 dBA.

2.3 Selection of Hearing Protectors

The Standard includes three methods of selecting HPDs based upon noise levels and the method the attenuation data has been obtained. In order of increasing potential accuracy, the methods are:

Use of classes, which pre-assigns the HPDs according to defined attenuation ranges

Use of single number such as NRR or SNR(SF84), and

Use of the octave bands approach.

Several tables in the document help the user select the best protector for a given noise environment.

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2.4 Specialized HPD

The Standard describes a new family of devices, intended to overcome some undesirable consequences of the use of the conventional HPDs, such as speech intelligibility and signal detection reduction, especially for people with hearing loss.

Those devices can be divided into two categories: active and passive. The active protectors make use of electronic circuitry to reduce the noise at the ear. In the passive protectors, the attenuation characteristics are altered using mechanical means.

Active protection devices included in the Standard are: noise-attenuating communication headsets, HPDs with direct music input, active noise reduction (ANR) and sound restoration devices.

Passive devices included are those with flat or uniform attenuation, frequency-sensitive protectors, adjustable-attenuation protectors and amplitude-sensitive (level-dependant) devices.

2.5 Fit, care and use of protectors

The Standard examines in details the importance of a proper fit and how to obtain it through proper training. The same applies to the way they are been taken care of, since keeping them in good shape is essential for obtaining the results that are expected.

2.6 Implementation of hearing protection devices

A section in the document is dedicated to the development and implementation of a hearing protection program that should include periodical training. It should also take care of posting of areas that may be hazardous and where the wearing of HPDs is compulsory.

Mention is made to the CSA Z1007 standard *Management of hearing loss prevention programs* that deals in details with those issues.

2.7 Filed Attenuation Estimation Systems (FAES)

FAES are devices used to test the attenuation obtained by an individual in a real situation. They intend to answer the fundamental question in hearing conservation practice: what amount of protection is a given individual really getting from his or HPD.

Fit testing reflects what a user can achieve and has been shown to achieve at the time of testing, and not necessarily what the user truly achieves day-to-day.

Fit testing systems may be used to help individuals to:

- Properly select their HPD
- Fit their selected HPD most effectively
- Learn consistency in the fitting of their HPD
- Spot-check the fit of their chosen HPD
- Estimate protected exposure levels

The result of testing using FAES is the so-called PAR (Personal Attenuation Rating). It is a single number that subtracted from the noise exposure of the worker estimates the noise exposure of the protected ear.

Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Relationship Between Direct And Indirect Methods Of Estimating Sound Exposure When Communication Headsets And Hearing Protectors Are Worn

Christian Giguère, Tim Kelsall

Individuals using communication headsets in the workplace are exposed to the external noise surrounding them as well as to the internal audio signals from the device. The 2013 revision of CSA Z107.56 introduced various methods of estimating the sound exposure from communication headsets that take into account both paths. Direct methods include specialized real-ear and simulated real-ear (i.e. manikin, artificial ear) measurement techniques. An indirect calculation method is also specified that only requires general sound measurement equipment. It is shown that the indirect method can be viewed an extension of the long-established calculation procedures for estimating the effective level under hearing protectors. In addition to the external sound level and the noise reduction rating of the device, the indirect calculation method for communication headsets also requires information on the effective listening signal-to-noise ratio of the audio channel. The new calculation method fills a gap by introducing a simple survey method for communication headsets that general practitioners and occupational health and safety professionals can use. This parallels the situation with hearing protectors where both calculation methods and more specialized measurement techniques (e.g. in-ear dosimetry under the device, field measurement systems) are available.

HEARING SCIENCES - SCIENCES DE L'AUDITION

THE IMPACT OF CONTEXT ON THE PERCEPTUAL ORGANIZATION OF SPEECH

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1 Introduction

One of the major functions of the human auditory system involves auditory scene analysis – separating a noisy environment into distinct auditory representations. Streaming is a well-studied laboratory task that is meant to model real-world object recognition, and typically involves the presentation of tonal stimuli in an ABA- type galloping pattern, with low A tones and high B tones[1]. When the frequency difference between A and B is small, participants typically perceive the tones as a single integrated source (ABA-ABA-…). When the frequency difference between A and B is large, participants tend to perceive two separate streams, known as streaming, with A repeating at twice the rate of B (A-A-A-A and B---B---…). At an intermediate frequency difference, perception is more ambiguous, though a series of recent studies found that both prior stimulus and prior perception have a large impact on how individuals classify ambiguous sequences [1,2].

Snyder and colleagues showed that when participants hear a non-ambiguous sequence (i.e., either one or two streams) followed by an ambiguous sequence, they are likely to classify the ambiguous sequence as the opposite of the initial sequence (impact of prior stimulus); for example, when presented with one stream followed by an ambiguous sequence, participants are likely to classify the ambiguous sequence as two streams. Interestingly, when two ambiguous sequences are presented in a row, the opposite effect is observed; if the first ambiguous sequence is classified as one stream, the next ambiguous sequence is more likely to also be classified as one stream (effect of prior perception [2]). Electrophysiological evidence suggests that the mechanisms through which recent stimuli/perceptual states influence the formation of auditory streams are at least partially independent from those influencing the current auditory stimuli [3].

Given the importance of verbal stimuli in auditory scene analysis, the objective of the current study was to extend the findings of Snyder and colleagues by using speech sounds instead of pure tones. Using more complex and ecologically valid stimuli, we gain a better understanding of auditory stream segregation and how we interact with the auditory world.

2 Methods

Sixteen healthy young adults (M_{age} = 23.25yr, SD = 4.39; 8 females) were recruited from the Baycrest participant database. All participants were right-handed except for one who was left-handed and all were fluent English-speakers with no known neurological or psychiatric issues and no history of hearing or speech disorders. Participants gave informed written consent according to guidelines established by Baycrest's Research Ethics Board.

Stimuli consisted of the vowel sounds /i/ (as in s*ee*) and /ae/ (as in c*a*t), henceforth referred to as "ee" and "ae". The vowels were presented in an ABA- pattern as ee-ae-ee-, with mainly first formant (f_1) frequency differences between A (ee) and B (ae). The stimuli represented one of three conditions: large f_1 difference between A and B (Δf_1 = 285Hz; typically perceived as two streams), small Δf_1 between A and B (Δf_1 = 47Hz; typically perceived as one stream), or intermediate Δf_1 (Δf_1 = 110Hz; ambiguous perception). Each trial consisted of an adaptation phase, which could have either a small, intermediate, or large Δf_1, as well as a test phase, which was always intermediate. Both phases were 7s each in duration, and were separated by a 1.44s inter-stimulus interval. In each phase, 14 repetitions of the ABA- triplets were presented sequentially, after which the participant made a response via button box indicating whether the previous sequence was perceived as one or two streams.

Participants were seated in a comfortable chair in a sound-attenuated chamber for the duration of the study. The testing session began with two hearing tests – the pure tone thresholds audiometry and the QuickSIN (speech-in-noise recognition). The order of the two tests was counter-balanced across participants. The concept of streaming was explained to participants and a brief practice session was given in order to familiarize participants with the stimuli and task. Participants were encouraged to keep their eyes fixated in a comfortable position and listen to the sounds. Participants completed five blocks of 30 trials each for a total of 150 trials, with each adaptation condition (small, intermediate, or large Δf_1) being presented 50 times throughout the study.

3 Results

During the adaptation phase, participants were more likely to report hearing two streams (streaming) when the formant difference between A and B was intermediate or large than when it was small (see Figure 1). There was also a difference in the ambiguous test trials based on which condition was presented at adaptation; participants were

[1] For an interactive example, please go to the following link: http://auditoryneuroscience.com/topics/streaming-galloping-rhythm-paradigm

significantly less likely to perceive intermediate test sequences as two streams as adaptation shifted from small to large Δf_1 [$F(2,14) = 10.479$, $p < 0.001$). These results demonstrate an effect of current/prior stimulus using speech sounds similar to previous psychophysical findings observed with tonal stimuli [1].

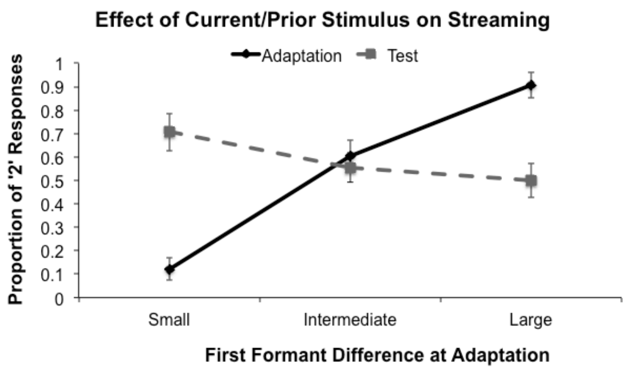

Figure 1. Effects of first formant differences on perception of streaming during the adaptation and test phase. Error bars represent standard error of the mean.

In order to investigate the impact of prior perception on subsequent classification, we looked at the likelihood of reporting streaming at test based on the perception of intermediate adaptation sequences. We found an increase in the perception of streaming at test when streaming was also perceived during intermediate adaptation sequences [$F(1,15) = 21.420$, $p < 0.001$; see Figure 2].

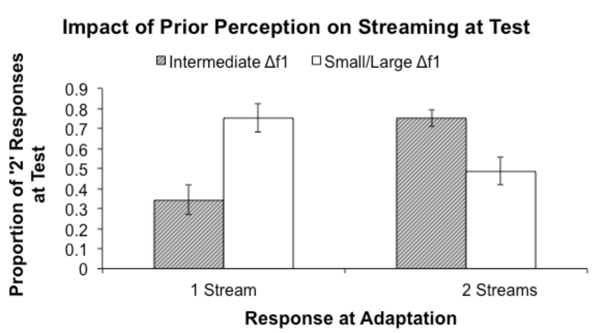

Figure 2. Effect of prior perception during adaptation on perception at test for ambiguous (intermediate Δf_1) and non-ambiguous (small/large Δf_1) adaptation sequences. Error bars represent standard error of the mean.

Importantly, this is the opposite of the prior stimulus pattern that was observed when the adaptation sequence had either a small or large Δf_1, as described above.

4 Discussion

Consistent with previous research using pure tones, we found that large differences in first formant between vowels (similar to frequency differences in tones) elicited more reports of streaming than a small difference between first formant transitions. Further, prior stimulus presentation seemed to bias current perception *away* from the stimulus that was just heard, while prior perception of ambiguous stimuli seemed to prime current perception *towards* the stimulus just perceived. Using tonal stimuli, Snyder and colleagues manipulated lags and intertrial intervals in order to precisely measure the time course of context effects, and concluded that it is unlikely that these effects are largely driven by response bias. Instead, it is suggested that different levels of neural representations reflect stimulus-related (i.e., Δf_1) and perception-related (i.e., 1 stream vs. 2 streams) processes, and that the effects of both processes build up over time [1,3]. Further research is required to determine whether similar mechanisms are responsible for speech sound segregation as tonal segregation, as well as whether the streaming of speech sounds is also affected by factors such as attention and knowledge [4].

One of the fundamental processes of the human auditory system is to organize sounds into meaningful elements, such as separating a police siren from the music playing through your car radio, or identifying and attending to your friend's voice in a noisy room. The findings of the current study support the notion that auditory stream segregation of speech sounds is impacted by context. A crucial next step would be to obtain electrophysiological data to better understand the neural mechanisms responsible for such processes. Also of interest would be the effects of aging on streaming of speech sounds, as previous research has shown age-related deficits in the sequential streaming of vowels [5].

Using complex, ecologically valid stimuli, we have replicated patterns of streaming previously only observed with pure tones. This study adds to the rich volume of literature characterizing the phenomenon of streaming and the effects of context on auditory scene analysis.

Acknowledgments

This research was supported by a Canadian Institutes of Health Research (CIHR) grant held by CA.

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MUSICIANS SHOW ENHANCED NEURAL SYNCHRONIZATION AT MULTIPLE TIMESCALES

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1 Introduction

Auditory events can be considered to have spectral energy at both short and long timescales. In the context of music, energy at short timescales determines the psychological phenomena of pitch, timbre, and sound-source localization, whereas long timescales give rise to the perception of pulse. Within the field of music cognition, there has been recent interest in neural oscillations and entrainment [1], whereby populations of neurons synchronize their firing with external stimuli, such as auditory events. Through this synchronization, it is thought that one is able to track periodic stimuli like tones and rhythms. Neural synchronization at both timescales can be measured using electroencephalography (EEG), and it has been found that the spectrum of the EEG signal closely matches that of the stimulus. Recent research has revealed that musicians, who in their daily lives are exposed to subtle differences in pitch and pulse, show better subcortical synchronization with energy at short timescales, as indexed by a larger peak at the fundamental frequency (f0) of stimuli [2]; a similar test for long timescales has not been investigated. In the current study, EEG was measured from musicians and nonmusicians while they listened to an isochronous sequence of tones. It was predicted that similar to subcortical synchronization at short timescales, the strength of cortical synchronization at long timescales would also increase with musicianship. Additionally, it was hypothesized that neural synchronization at multiple timescales would successfully predict measures of musical experience in a regression analysis.

2 Methods

Participants (musicians, amateur musicians, and nonmusicians) were seated in a soundproof booth and fitted with foam insert earphones for the duration of the experiment. Participants first completed behavioural tasks assessing pitch discrimination and pulse perception before being fitted with an EEG electrode cap. Participants listened to an isochronous sequence containing periodicities at short timescales (fundamental frequency of each tone) as well as long timescales (the rate of presentation of tones). Each tone had a fundamental frequency of 220 Hz and energy at the first 5 harmonics (440, 660, 880, 1100, and 1320 Hz). Tones were presented at a rate of 2.5 Hz (period of 400 ms) with an intensity accent on the first tone of every group of 4 (0.625 Hz). Afterwards participants completed questionnaires assessing their history of playing music, listening habits, and current musical activities.

Subcortical and cortical activity were recorded on separate systems and at different sampling rates to accommodate for the different timescales present in the stimulus. Subcortical activity was recorded at 20000 Hz using earlobe electrodes and a forehead ground. Data were bandpass filtered between 100 and 1500 Hz, the response to each tone was extracted as a separate trial, and trials exceeding 35 μV were discarded as artifacts [3]. Trials were averaged together and a fast-Fourier transform (FFT) of the resulting waveform was calculated with a bin size of 5 Hz. The spectrum of a pre-stimulus baseline period was subtracted to correct for the noise floor [4].

Cortical activity was recorded at 512 Hz with a 64-channel electrode cap. Data were highpass filtered at 0.1 Hz and subjected to an independent components analysis (ICA) to identify the presence of eyeblink and other artifacts. The contribution of artifactual components was removed from the timeseries data. Data were segmented into epochs containing 64 tones (approximately 25 seconds) and an FFT with a bin size of 0.0312 Hz was calculated for each electrode [1]. The contribution of the noise floor was removed by subtracting, from each frequency bin, the average of two bins on either side lying two bins away from the current bin [1]. The resulting spectra were averaged across all electrodes.

3 Results

Average spectra for subcortical synchronization to tones can be found in Figure 1. Musicians showed greater synchronization strength to the fundamental frequency of the tone, but this difference was only marginally significant ($t(8)=2.010$, $p=0.079$).

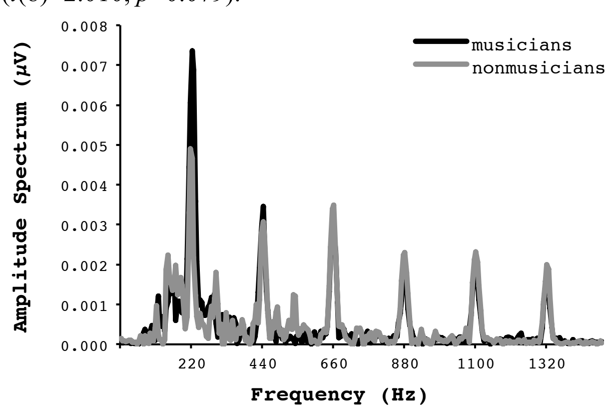

Figure 1: Spectrum of subcortical response to the 220 Hz tones. Synchronization strength at the fundamental frequency is larger for musicians as compared to nonmusicians.

Average spectra for cortical synchronization to the pulse can be found in Figure 2. Musicians showed greater synchronization strength to the pulse frequency, but this difference did not reach significance ($t(8)=0.884$, $p=0.402$).

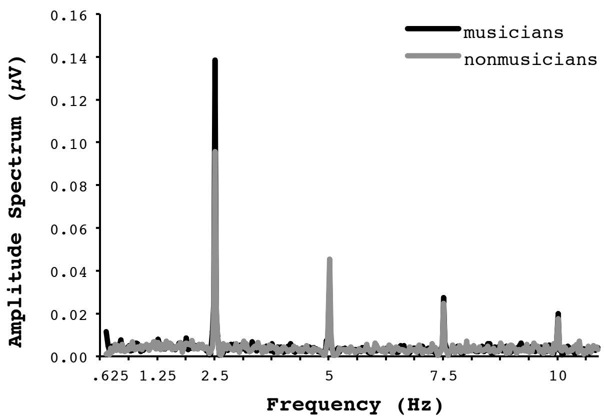

Figure 2: Spectrum of cortical response to the 2.5 Hz pulse rate. Synchronization strength at the fundamental frequency is larger for musicians as compared to nonmusicians.

The extent of synchronization to both timescales did not correlate with years of musical experience, however they did correlate with measures of current musical engagement. Specifically, subcortical synchronization to the fundamental frequency of tones significantly correlated with the number of hours spent playing music ($r=0.55$, $p=0.021$), and cortical synchronization to the pulse frequency was marginally correlated with the number of hours spent listening to music ($r=0.46$, $p=0.067$).

A regression analysis (Figure 3) on the total number of hours spent playing or listening to music, using neural synchronization strength to the fundamental frequency of the tone and pulse frequency, explained approximately 31% of the variance and was marginally significant ($R^2=0.314$, $p=0.071$).

4 Discussion

The current study investigated differences in neural synchronization between musicians and nonmusicians. It was found that while musicians yielded larger spectral peaks than nonmusicians at the frequency of the stimulus at both short and long timescales, these differences were only marginally significant. Although the strength of peaks did not correlate with years of musical experience as predicted, they did correlate with the number of hours per week spent playing or listening to music. Previous research indicates that musical experience relates to the strength of subcortical synchronization [2, 3], but the current study suggests a more nuanced explanation. The amount of time currently spent engaged with music might be a better predictor of neural synchronization. Additionally, the strength of neural synchronization at short and long timescales together can predict the amount of time an individual spends engaged in musical activities. This kind of coherence in synchronization across timescales and levels of musicianship is a novel finding and suggests that synchronization abilities in different parts of the brain may be improved concurrently with musical engagement. Since music typically contains nested periodicities, it makes sense that an ability to faithfully represent these periodicities neurally would be necessary for listening or playing.

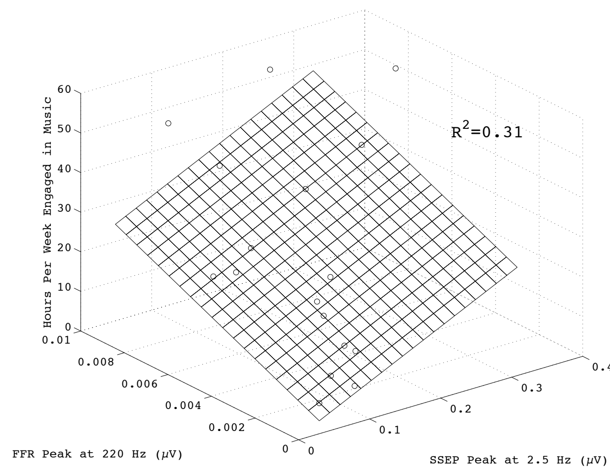

Figure 3: Plot of regression analysis. The size of subcortical frequency-following response (FFR) peaks to tones and the size of cortical steady-state evoked potential (SSEP) peaks together may predict the number of hours an individual is engaged with music.

5 Conclusion

These findings indicate that the experience-dependent plasticity observed in musicians manifests itself at multiple cortical levels corresponding to oscillations at different timescales present in music.

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Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Impact Of Auditory Attention On The Efferent Auditory System In The Absence Of Real Auditory Targets.

Wing Yiu Stephanie Wong, Aravind Kumar Namasivayam, Pascal H. H. M. Van Lieshout

Previous studies have compared visual and auditory attention to no task conditions and have demonstrated an attention driven modulation of the efferent auditory system (De Boer & Thornton, 2007; Maison et al., 2001). However, it is unclear whether these effects are modality-specific or a result of a generalized attentional processes. In the present study, 16 young adults observed facial speech gestures related to productions of vowels /a/ and /u/ in the presence of contralateral noise under two instructions: (a) visually count the number of /a/ productions and ignore contralateral noise (visual attention) and (b) to listen carefully and detect target sound /a/ embedded in contralateral noise (sham condition; auditory attention). These "œsham" trials did not have any acoustic targets and investigated the effect of auditory attention even when there was no real target. The influence of visual and auditory attention on the efferent auditory system were indirectly assessed by examining the effects of Contralateral Suppression on Transient-evoked otoacoustic emissions (CS-TEOAE paradigm; Collet et al., 1990). The Mean (S.D.) change from baseline for visual attention and auditory attention were 2.19 (1.98) and 1.88 (1.82), respectively. Cohen's d for the mean difference between the two conditions yielded a moderate positive effect size = 0.52. 12 out of 16 (75%; exact binomial test significant at one tailed p = 0.03) participants demonstrated a greater suppression of TEOAEs (mean difference = 0.31 dB SPL) in the sham/auditory attention condition relative to the visual attention condition. These effects are similar to those reported in the literature, wherein attention to stimuli in the contralateral ear increased OAE suppression (Harkrider & Bowers, 2009). Our results show that these effects are obtainable even in the absence of real auditory targets (i.e. without stimulus confound). Overall, these findings suggest a modality-specific attentional modulation of the efferent auditory system.

PHYSICAL ACOUSTICS / ULTRASOUNDS - ACOUSTIQUE PHYSIQUE / ULTRASONS

Control of Acoustic Resonance in Shallow Rectangular Cavities Using Surface Mounted Blocks

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1 Introduction

Flow over cavities has been identified as a potential source of acoustic resonance in many engineering applications [1, 2]. The inherent instability of the shear layer over a cavity can give rise to pressure oscillations. A feedback mechanism can result in sustained oscillations in the flow field [3]. The oscillations of the shear layer at the cavity edge cause an oscillating mode at frequencies determined by, $St_n = f_n l/U = 0.5n, n = 0, 1, 2, \ldots$,where l is the characteristic shear-layer length. St_n is the Strouhal number corresponding to the frequency f of the n^{th} mode at a flow velocity U [4].

At flow velocities where a shear-layer mode frequency coincides with one of the acoustic cross-modes, a look-in may occur, producing a tonal noise of extremely high sound pressure levels. The suppression of the cavity noise using passive devices that distort the shear layer is investigated in the literature [5, 6]. However, the nature of the interactions between an upstream flow disturbance with the shear layer over the cavity mouth on the acoustic resonance mechanism is not fully understood.

A surface-mounted block in the flow is known to cause a complex pattern of vorticity downstream. Hussein and Martinuzzi experimentally studied the channel flow around square cross-section surface-mounted blocks [7]. They found that the complex features of the flow are highly dependent on the width of the block. Wider blocks caused higher turbulence intensity and longer wake region downstream. They also noted that the vorticity field near the cube was characterized by the generation of the hairpin vortices in the near-wake region with relatively low frequencies, and the shedding of lateral vortices from the leading lateral edges of the cubic obstacle.

The objective of this study is to investigate the effect of the attachment of square cross section blocks of different widths at different distances from the shallow cavity upstream edge on the intensity of the resonance resulting from the fluid-resonant interactions of the air flow over the cavity mouth.

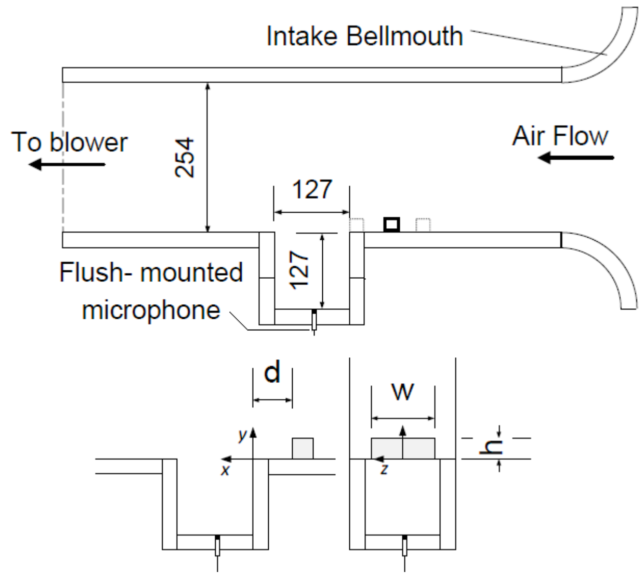

Figure 1: Test section configuration and the used nomenclature.

2 Experimental setup

The experiments are carried out in an open loop wind tunnel. The dimensions and the used nomenclature for the experiments are shown in figure 1. The cavity depth, length, and width are equal to 0.127 m. The channel height is 0.381 m, which imposes a first acoustic cross-mode with a frequency f_1=465 Hz. The response of the system is characterised by the dimensionless pressure, $P^* = P/(0.5\rho U^2)$, and the flow velocity is indicated in the reduced form $U_r = U/f_1 l$, based on the frequency of the lowest cross-mode of the test section.

Six Blocks of square cross-section with edge length h=0.0191 m are used, with width to height ratios w/h=1, 2, 3, 4, 5, and 6.66. The blocks are attached at different locations upstream of the cavity leading edge with d/h = 0, 3, and 6. These locations are selected to investigate the effect of the reattachment position of the lateral flow around the block sides.

The cavity is tested at air velocities up to 160 m/s. A flush-mounted microphone located at the center of the cavity bottom is used to measure the acoustic pressure. The resonance frequency and sound pressure level are recorded and analysed at each flow velocity.

3 Results

3.1 Cavity response with no blocks

The acoustic response of the cavity in cross-flow provides a base case on which the control effectiveness is assessed. The characteristics of the noise in the cavity and the shear layer modes that excite the resonant sound are identified.

Figure 2 shows the response of the cavity with no block attached. The upper part of figure 2 shows that the peaks of the dimensionless pressure occur at the coincidence of the shear-layer modes with the acoustic cross-modes. The first cross-mode with f_1=465 Hz peaks at St=0.48, 1.11, and 1.61 referring to coincidence with the first, second, and third shear-

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layer modes ; respectively. Development of the normalized pressure with reduced velocities is shown in the lower part of figure 2. The pressure reached P^*= 0.01 at U_r=0.55 for the third shear-layer mode, P^*=0.12 at U_r=0.95 for the second shear-layer mode, and P^*=0.27 at U_r=2.10 for the first shear-layer mode.

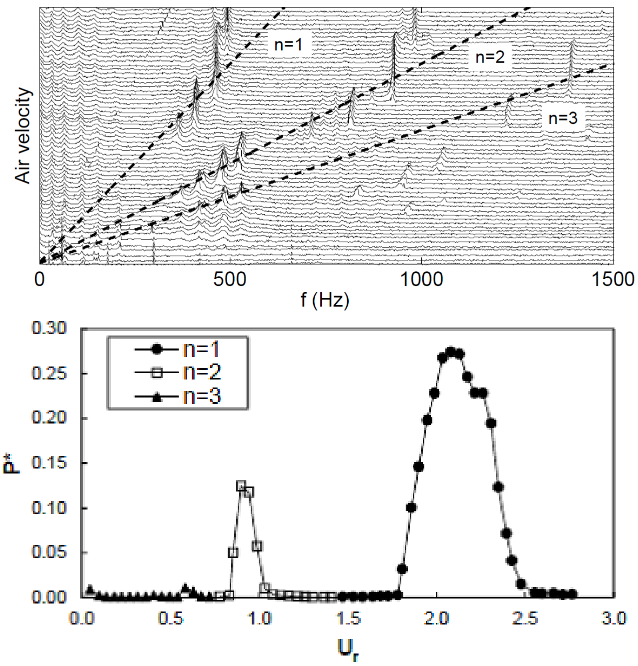

Figure 2: Response of the cavity with no block.

3.2 Effect of block attachment

Figure 3 shows the attenuation obtained using different configurations of the blocks at different upstream locations. Results show that the cases with moderate values for width and distance are the most effective in suppressing the acoustic resonance excitation. Attenuation values of 30 dB are obtained using blocks attached at a distance of d/h=3 with blocks of widths w/h of 3 and 4. Cubic blocks cause less than 5% attenuation levels. Blocks that filled the whole section of the wind-tunnel channel negatively affect the sound pressure level, increasing the noise. These results suggest that the horseshoe vortices that the block induces at its vertical upstream sides are important for noise suppression.

4 Conclusions

A passive method for controlling the acoustic resonance resulting from subsonic flows with Mach numbers up to 0.45 over shallow rectangular cavities is investigated. Experiments are performed to investigate the effectiveness of attaching blocks of square cross-section in suppressing the flow-excited acoustic resonance in shallow rectangular cavities. Six different square blocks with width to height ratios of w/h = 1, 2, 3, 4, 5, and 6.66 are investigated. The blocks are attached at different locations upstream of the cavity leading edge with

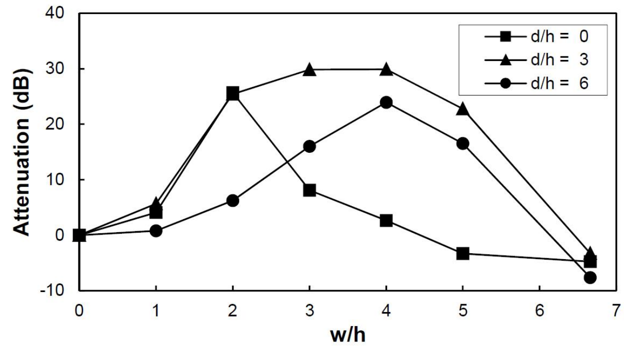

Figure 3: Attenuation of the cavity noise using different block configurations

d/h = 0, 3, and 6, where d is the upstream distance from the cavity leading edge and h is the block height.

The results show that significant attenuation of the generated acoustic pressure with up to 30 dB is achieved using blocks of w/h between 2 and 4. Moreover, it is observed that the most effective attenuation of the acoustic resonance is achieved when the blocks are located at a distance of $3h$ upstream of the cavity leading edge. Blocks with moderate widths are more effective than blocks that fills the whole width of the wind tunnel.

Acknowledgement

The authors thankfully acknowledge the financial support provided by the Natural Sciences and Engineering Research Council of Canada (NSERC).

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LOCALIZATION OF ACOUSTIC EMISSION SOURCE IN PLATES USING WAVELET TRANSFORM

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1 Introduction

AE techniques for crack detection, leak and loose part monitoring in different engineering applications are gaining much interest due to their potential capabilities for in-service inspection without the need of operational shutdown. However, many issues remain challenging to implement this technique efficiently in industrial setting.
This work develops and implements an Acoustic Emission (AE) source localization technique based on wavelet transform analysis (WT) and modal acoustic emission (MAE) [1] and compares the accuracy of measuring the Time of Arrival (TOA) of S0 and A0 modes for both minimum value of wavelet coefficient [2] and the maximum peak value [3]. Moreover, time-frequency analysis was conducted and the group velocity dispersion curves were overlaid to clearly identify the modes of the acoustic emission (AE) signal and locate its source. The effect of the plate size on the developed technique is also investigated.

2 Theoretical background

All source localization techniques mainly depend on determination of the time of arrival for the AE signal at different arranged sensors. There are two ways to measure TOA: traditional AE methods and alternative methods. Traditional AE methods assume threshold value and consider the arrival time is at first hit crossing the threshold [4]. The alternative method, single sensor modal analysis location, depends on MAE theory. This method is based on of lamb theory. As lamb wave propagating in the plate has dispersive characteristics time deference between arrival of the fundamental modes S0 and A0 at the sensor could lead to the source location [1, 5]. As shown in figure (1) the velocity of these two modes can be determined from dispersion curves at certain frequency.

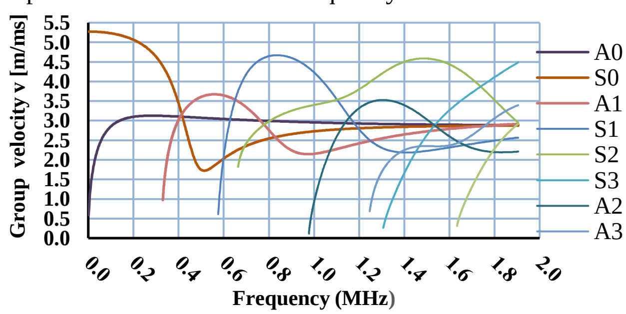

Figure (1) Dispersion curves for Steel plate 4.76 mm thick.

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The source to sensor distance (L) can be calculated by this eq. [2],

$$L = \Delta t(V_{S0}V_{A0}/V_{S0-}V_{A0}) \quad \text{.......... (Eq. 1)}$$

Where: Δt is the time difference between the arrival of S0 and A0 modes, V_{S0} and V_{A0} are their velocity

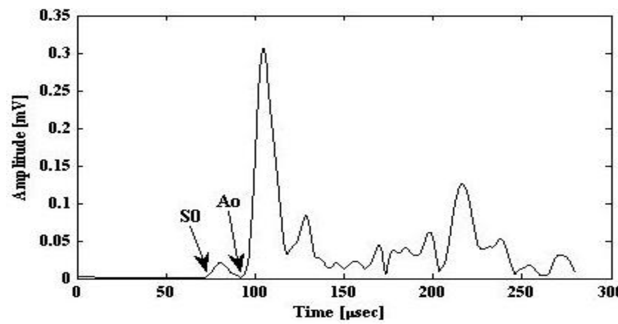

Figure (2) Relation between wavelet coefficient and time

Figure (2) shows the relation between wavelet coefficient and time for the transformed waveform at certain frequency (271 kHz). Hamstad and Jeong [3, 7] stated that time difference between the arrival of S0 and A0 modes can be measured by using the peak value of wavelet coefficient however Shukri [2] stated that the minimum value of wavelet coefficient represents the actual arrival time of these modes.

3 Experimental set-up

Figure (3) The experimental set-up for the steel plate

A pencil-lead break test was carried out for two plates with different materials and geometrical configurations. An aluminum plate with dimensions 100mm×100mm and 3mm thick was tested to represent relatively small plate. However, a mild steel plate with dimensions 1000mm×800mm and 4.76mm thick considered as relatively larger plate.

Figure (3) represents the experimental set-up for the steel plate. KISTLER 8152B1 sensor with working frequency range of 100-400 kHz were used throughout this work. The sensor was connected to KISTLER AE-piezotron coupler type 5125. The sensor was coupled to the surface of the plate with high vacuum white grease.
A four channel NI DAQ (PCI – 6132) was used to acquire the AE waveform. The acquired waveforms were then stored on a computer for further analysis.

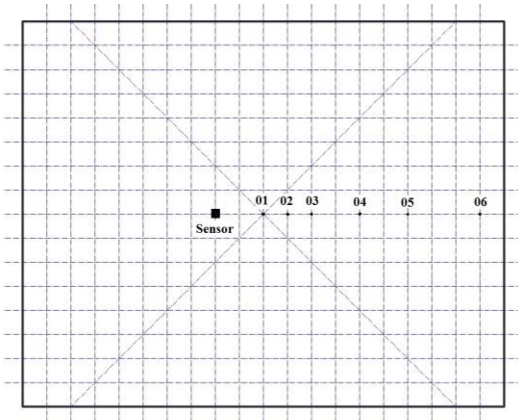

Figure (4) Location of sensor and points of pencil lead break points (grid density is 50 mm)

Five Hus-Nielson (H-N) sources were conducted at distance of 100, 150, 200, 300, 400 and 550mm on the top surface of the steel plat as shown in figure (4).

4 Results and desiccations

Figure (5) shows a typical result of WT of the detected signal via AGU-Vallen wavelet software [6]. It is clearly shown in this figure that S0, A0 and A1 are easily distinguished and the accurate arrival of S0 and A0 can be determined.
Average value for source to sensor distance was calculated at different points by using both maximum and minimum values in measuring S0 and A0 modes arrival time and compared with actual distance. The absolute percentage of detection error between the actual and calculated distance was determined.

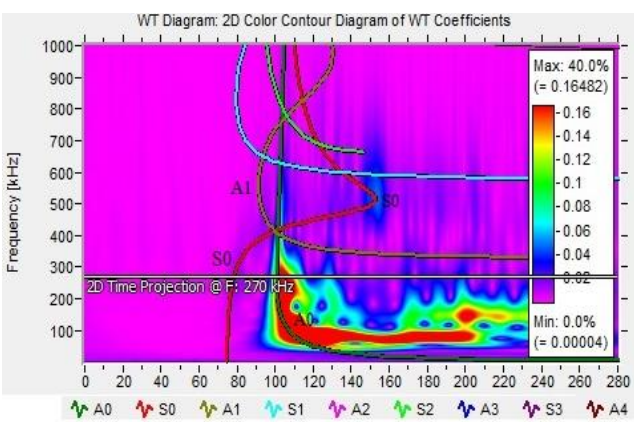

Figure (5) Typical result of WT of the detected signal

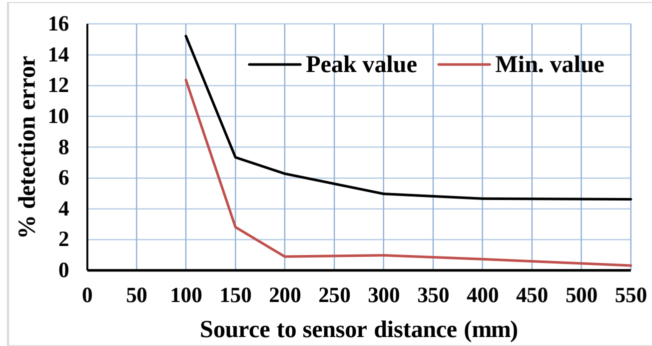

Figure (6) Source to sensor distance against % detection error for both peak value and minimum value.

Relation between source to sensor distance against percentage detection error for both peak value and minimum value is shown figure (6).

5 Conclusions

This paper presents an implementation of Acoustic Emission (AE) technique to detect an acoustic emission source in plates with different materials and geometrical configurations

A. For relatively smaller plates
- The results show difficulty in the use of wavelet transform to determine the source location due to reflection of the signal at the plate side edges.

B. For relatively larger plats
- Source location shows an acceptable accuracy by using wavelet transform
- The technique for extraction of the arrival time through the minimum wavelet coefficient value is more accurate rather than the peak value.
- Further work aims to apply this technique for a steel pipe under fatigue stress.

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Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Acoustical Investigations Of Molecular Interactions In Polymer Solutions Of Pan/clay Nano Composites And Dmso

Devinder Pal Singh, Arun Upmanyu

Acoustical studies of intermolecular interactions in the polymer solution of PAN / clay nanocomposites and DMSO have been done at 30 degree Celsius using experimental ultrasonic velocity and density data taken from literature. Several acoustical and thermo-dynamical parameters such as isothermal compressibility, adiabatic compressibility, specific heat ratio, volume expansivity, surface tension, specific sound velocity, specific adiabatic compressibility, intermolecular free length, pseudo-Gruneisen parameter and classical absorption coefficient have been evaluated. Some elastic parameters such as Young modulus, shear modulus, bulk modulus and Poisson ratio have also been determined. Non linear parameters such as Moelwyn-Hughes parameter, reduced volume, reduced compressibility, Sharma's constants, Huggins parameter, isobaric acoustical parameter, isochoric acoustical parameter, isothermal acoustical parameter, fractional free volume, repulsive exponent, thermo acoustical parameter such as A*and B*, Bayer's non-linear parameter, internal pressure, isochoric thermo-acoustical parameter and isochoric temperature coefficient of internal pressure have also been calculated. The Moelwyn-Hughes parameter has been utilized to establish relation between the Bayer's non linear parameter, internal pressure and Sharma constant. Relationships among the isobaric, isothermal and isochoric thermo- acoustical parameter have been studied and analyzed for PAN/clay nano composites. The obtained results have been compared with the experimental results as available in literature. The non-ideal behavior of the polymer solution has been explained in terms of its composition and variation of its acoustical and thermo-dynamical parameters. The present treatment offers a convenient method to investigate thermo-acoustic properties and anharmonic behavior of the system under study.

SPEECH SCIENCES - SCIENCES DE LA PAROLE

PERCEPTION OF INTERVOCALIC CONSONANT CLUSTERS BY JAPANESE LISTENERS

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1. Introduction

Perceptual epenthesis by Japanese listeners has been known as a quintessential example of phonological bias in speech perception. Since Japanese has a phonotactic constraint that prohibits the occurrence of non-geminate consonant clusters, it has been argued that Japanese listeners upon hearing such consonant clusters tend to perceive an illusory vowel /u/ between the consonants so that their percepts conform to the phonotactic constraint. Experimental studies have demonstrated that because of this perceptual epenthesis Japanese listeners have significant difficulty in perceiving the difference between stimuli that contain a non-geminate consonant cluster (e.g. ebzo) and their counterparts with an intervening /u/ (e.g. ebuzo) [1].

The prohibition of non-geminate consonant clusters, however, is not absolute in Japanese. Some clusters are actually found in the surface phonetic form of naturally produced utterances as a result of vowel devoicing, which is usually described as the high vowels /i, u/ being devoiced when they occur between voiceless consonants. However, the actual outcome of the process ranges from devoicing to deletion [2], and its occurrence is rather probabilistically determined by various phonological factors, such as vowel type (e.g., high vowels are more likely to be devoiced than non-high vowels) and consonant type (e.g., devoicing is more likely to occur next to voiceless consonants than voiced consonants) [3].

Since vowel devoicing is an allophonic rule, it is assumed that once Japanese listeners learn it they should be able to recover underlying vowels whenever they hear consonant clusters that may have arisen through vowel devoicing. However, studies have demonstrated that it is not the case. Recovery does not happen unless it is lexically supported. In other words, Japanese listeners seem to retain the phonetic representation of surface consonant clusters if their lexical knowledge does not motivate the recovery of underlying vowels [3]. This is further supported by a finding that Japanese listeners actually have access to the phonetic representation of surface consonant clusters in a speech perception task. When they were asked to rate the goodness of non-word stimuli with surface consonant clusters (phonotactically non-canonical but phonetically natural stimuli) and their non-word counterparts with an underlying vowel (phonotactically canonical but phonetically unnatural stimuli), they rate the former as better than the latter [4]. This suggests that Japanese listeners have phonetic knowledge about surface consonant clusters.

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The goal of this study is to understand how Japanese listeners' phonetic knowledge about surface consonant clusters relates to perceptual epenthesis. It examines whether the perceptibility of non-geminate consonant clusters for Japanese listeners is affected not only by their phonotactic knowledge about the prohibition of those clusters, but also by their phonetic knowledge about how likely those clusters are to occur as a result of vowel devoicing. In order to do that, 19 Japanese listeners were tested on discrimination between non-geminate consonant clusters and their counterparts with an intervening /u/. Crucially, 4 different types of consonant clusters with different degrees of likelihood of occurrence as a result of vowel devoicing were tested. If perceptual epenthesis is solely triggered by phonotactic knowledge, there would be no effect of cluster types.

2. Methods

2.1. Design

An AX discrimination paradigm was used. In experimental trials, participants compared non-word stimuli with intervocalic non-geminate consonant clusters (VCCV) and their counterparts with an intervening /u/ (VCuCV). Consonant clusters were classified into 4 types according to the likelihood of their occurrence as a result of vowel devoicing; from lowest likelihood to highest, these were voiced stop - voiced stop (DD), voiceless stop - voiced stop (TD), voiceless stop - voiceless stop (TT), and voiceless fricative - voiceless stop (ST) [3] (Table 1). In filler trials, participants compared stimuli with a mid short /u/ (VCuCV) and their counterparts with a mid long /u:/ (VCu:CV).

Cluster type	Stimuli	Likelihood
DD	igdo, egba, ibgo, ebda	Low
TD	ikdo, ekba, ipgo, epda	Mid-low
TT	ikto, ekpa, ipko, epta	Mid-high
ST	isto, espa, isko, esta	High

Table 1: Cluster types.

2.2. Stimuli

A trained phonetician whose native language is Japanese produced stimuli with a mid short /u/ (e.g., igudo). Stimuli were normalized in vowel duration and intervocalic interval duration. The mid vowel was spliced out to generate stimuli with consonant clusters (e.g. igdo) or lengthened to generate stimuli with a long /u:/ (e.g. igu:do).

2.3. Procedure

In each trial, participants heard a pair of stimuli and decided whether the stimuli were two different things or two repetitions of the same thing. The ISI was set to 500 msec. A block consisted of 64 experimental trials (32 same trials and 32 different trials) and 64 filler trials (32 same trials and 32 different trials). The block was repeated twice with an intervening short break.

3. Results and discussion

Performance was measured in terms of sensitivity (d prime) separately for filler trials (Figure 1) and experimental trials (Figure 2).

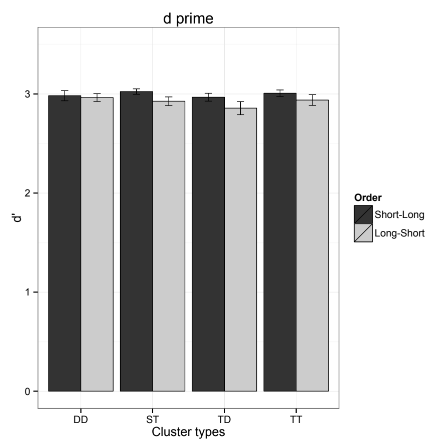

Figure 1: d prime scores for filler pairs

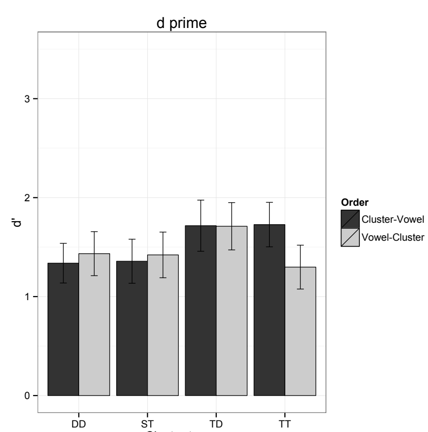

Figure 2: d prime scores for experimental pairs.

Cluster type and order of stimuli presentation were taken into analyses. D prime scores for the filler trials show that Japanese listeners have good sensitivity to the vowel duration contrast. A repeated measures ANOVA revealed no main effect or interaction. D prime scores for the experimental trials show that Japanese listeners have poor sensitivity to the contrast between consonant clusters and their counterparts with an intervening /u/. A repeated measures ANOVA revealed a main effect of cluster type ($F(3, 54)=2.77$, $p<0.05$); d prime scores were numerically lower for DD and ST cluster types than TD and ST cluster types. The difference, however, did not follow the order of likelihood of occurrence. Interestingly, Japanese listeners had more difficulty in perceiving clusters of the least likely type (DD) and the most likely type (ST). This can be potentially explained by biases coming from two different kinds of phonological knowledge, one about the phonotactic constraint and the other about the allophonic rule. Clusters of the DD type were hard to perceive due to perceptual epenthesis, and clusters of the ST type were hard to perceive due to the recovery of the underlying /u/. Clusters of the TD and TT types were relatively easier to perceive. This is probably because these types are relatively immune to the phonological biases. The TD and TT types are more frequent than the DD type and do not trigger perceptual epenthesis as strongly as the DD type does, while the TD and TT types are less frequent than the ST type and do not trigger the recovery of underlying /u/ as strongly as the ST type does.

The ANOVA also revealed a marginal interaction between cluster type and order of stimuli presentation ($F(3, 54)=2.497$, $p=0.0694$). A post-hoc analysis revealed that for the TT type, d prime scores were significantly higher when stimuli were presented in the order of VCCV and VCuCV. Previous studies have argued that in an AX paradigm discrimination is relatively easier when stimuli are presented in the order of unfamiliar and familiar items [5]. If the ordering effect is interpreted in terms of stimulus familiarity, what defines familiarity in this context is phonotactic canonicality.

4. Conclusion

The results showed that the perceptibility of non-geminate consonant clusters varied depending on cluster type, suggesting that the phonotactic constraint is not the only factor that affects the perception of non-geminate consonant clusters. A further study is needed to understand the mechanism behind the processing of non-geminate consonant clusters by Japanese listeners.

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Stop consonant production by French-English bilingual children in Southern Alberta

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1 INTRODUCTION

Although Canada has two official languages, French and English, it is well-known that these languages are not used equally across the country. Quebec is French-dominant, with nearly 8 out of 10 people reporting French as their native language compared to 8.3% who are native Anglophones (Statistics Canada, 2012b). In Alberta, English is the mother tongue of 77.0% of residents, with native Francophones making up only 1.9% of the population (Statistics Canada 2012a). While quite a few studies have investigated French-English bilingualism in Francophone Canada (e.g. Caramazza et al., 1973; Fowler et al., 2008; MacLeod & Stoel-Gammon, 2008), there exists a dearth of research on this topic in Western Canada where the sociolinguistic context is the reverse. We are interested in exploring how the differing social context might impact French-English speaking children's speech production. Thus, our study is set within the city of Lethbridge in southern Alberta—where the native French population is only 0.9% (Statistics Canada, 2012a). Official language status has helped maintain Franco-Albertan culture in the region, however, and the city boasts a French public library, community centre, and K-12 school, among other services. (ACFA regionale de Lethbridge - Medicine Hat, n.d.). The present study examines French and English speech sound production in 10 grade five students attending a Francophone school in Lethbridge using analysis of voice onset time (VOT).

VOT is an acoustic parameter that measures the time lapse between stop consonant release and the voice onset of the following vowel (Lisker & Abramson, 1964). English and French both have distinct stop consonant sets that can be measured and compared using VOT: the voiced /b/, /d/, and /g/, and the voiceless /p, /t/, and /k/. Monolingual English- and French-speakers differ in VOT measures reported for both sets, with Anglophones producing long-lag VOT (>30 ms) for voiceless stops and short-lag VOT for voiced stops (<30 ms) (Docherty, 1992). By contrast, Francophones' voiceless stops are in the short-lag VOT range, while their voiced stops exhibit negative VOT values indicative of glottal action preceding the stop burst (i.e., prevoicing) (Ryalls & Larouche, 1992). French-English bilinguals, who are the focus of this study, often exhibit different VOT patterns altogether. For example, Flege (1987) demonstrated that the English voiceless /t/ is shorter (i.e., more French-sounding) in native English-speakers who have learned French. Moreover, Fowler et al. (2008) found that adults who learned both French and English before age 3 present voiceless VOT values in both languages that are influenced bidirectionally. In other words, each language system is influenced by the other - even though a clear distinction is made between the two. Bidirectional language influence has also been demonstrated in early Korean-English bilinguals (Baker & Trofimovich, 2005).

To our knowledge, no previous study has employed VOT to study speech sound production in Albertan French-speaking children. Our research seeks (1) to determine how the VOT patterns of French-speaking children in Anglo-dominant Lethbridge compare to those of native Quebecois and monolingual English speakers and (2) if they have developed separate language systems for English and French.

2 METHOD

A total of 10 grade five students (aged 9-11) took part in the study. All participants were French-English bilingual speakers enrolled in a local Francophone school in Lethbridge, Alberta.

A word repetition task was employed to assess participants' VOT production patterns of word-initial voiced and voiceless stop consonants in both French and English. Audio prompts consisted of pre-recorded natural speech from a monolingual English speaker for the English component and a native French speaker for the French component. Six initial stops were examined (/p/, /b/, /t/, /d/, /k/, /g/) immediately followed by one of three consistent vowel environments /i/, /u/, /æ/ (English) or /i/, /u/, /a/ (French). During the task, the participant was sat in front of a computer monitor in a quiet room where visual stimuli were presented one by one on the computer screen, accompanied by a matching auditory speech token. The computer program Show & Play (Edwards & Beckman, 2008) was used to couple auditory and visual stimuli. Prior to engaging in the task, the student was provided with verbal instructions where they were asked to repeat each word back into the microphone after it had finished playing over the speakers.

Participants produced 9 words for each stop in both English and French for a total of 54 words in each language. Children's speech production was recorded using a Marantz flashcard recorder (model: PMD661). The speech stream was then segmented and events such as bursts and voice onset were marked using Praat (Boersma & Weenink, 2013) to permit extraction of VOT for acoustic analysis.

3 RESULTS

Table 1 displays the mean VOT values of the French-English bilingual students examined in this study. In addition, the values of monolingual English and Quebec-French speaking children are reported to serve as a comparison (Netelenbos, 2013; Ryalls & Larouche, 1992). When producing French voiced stops, the students are not prevoicing to the same extent as the extreme negative VOT values produced by native French speakers in Quebec. In the bilingual students' English productions, the voiced stops are not too far off the mark when compared to monolingual English children, with the exception of /b/, which is prevoiced by the French-English bilinguals. In their voiceless stops, the bilingual children produce VOT values appropriate for each language and consistent with the two language comparison populations. Next, it was of interest to determine whether the bilingual students' two language systems are behaving independently of each other. Upon visual inspection of the bilinguals' voiced and voiceless stop productions, it can be observed that the voiceless stops /p/, /t/, and /k/ exhibit large VOT differences between English and French, in that their duration is considerably longer in English. Conversely, the voiced stops are rather similar in both languages. This observation was confirmed by a two-way repeated measure ANOVA (language * stop) that revealed a significant interaction between the two variables, suggesting these children only maintain two language systems for certain stop sounds. A subsequent repeated measure ANOVA for each stop consonant indicated a significant language effect for the three voiceless stops, /p/, /t/, and /k/, ($p<0.001$ for all three stops), but was absent for the three voiced stops ($p=0.07$ for /b/, and $p>0.1$for /d/, and /g/).

	Bilingual English	Bilingual French	Mono-English	Quebec French
/p/	69 (22)	39 (36)	74 (27)	32 (12)
/t/	75 (23)	53 (44)	81 (20)	60 (22)
/k/	88 (27)	59 (43)	89 (25)	65 (15)
/b/	-18 (76)	-3 (56)	8 (30)	-91 (24)
/d/	-6 (53)	-12 (60)	-5 (60)	-91 (21)
/g/	3 (56)	-8 (67)	16 (48)	-88 (28)

Table 1: Mean VOT values (in milliseconds) of French-English bilingual children in Lethbridge (9~11 years old); monolingual English children in Lethbridge (9 years old; Netelenbos, 2013); and native French children in Quebec (9~11 years old; Ryalls & Larouche, 1992). Standard deviations are provided in brackets.

4 DISCUSSION

The results of the present study suggest that the English-dominant setting in Lethbridge has an anglicizing effect on bilingual French-English speakers' French system. Compared with Francophones in Quebec (Ryalls & Larouche, 1992), Lethbridge bilinguals exhibit less negative VOT values for voiced stops, but are similar with regard to voiceless stops. The prevoicing pattern in French voiced stops requires greater vocal exertion than in English, thus French-English bilinguals in Lethbridge may be unconsciously shifting their French voiced stop set to the easier-to-master English pattern. In a longitudinal study of seven months, Simon (2010) studied the prevoicing frequency of a native Dutch-speaking child who had moved to the United States with his family. Over the course of the study, the child gradually produced fewer prevoiced VOT values in his native language as his length of exposure to the Anglo-dominant sociolinguistic setting increased. While our study is not longitudinal, we believe it provides similar insight into the environmental effects on language development in bilingual children.

Despite living in predominantly Anglophone Lethbridge, bilinguals are exhibiting moderately prevoiced values for /b/ in English. This may indicate that their French is having a gallicizing effect on their English. Sundara, Polka, & Baum (2006) found that French-English bilinguals living in Montreal produce VOT values for /d/ in English that are significantly negative; however their study took place in French-dominant Quebec, thus the sociolinguistic input is different from ours. It is important to note that there was some demographic variance within our participant sample, with 4 simultaneous bilinguals since birth, 3 native French-speakers who learned English at age five or older, 2 trilinguals with native languages of Spanish and Dutch, and 1 Anglophone who learned French at age five. This heterogeneity may explain the high standard deviation for the English /b/. Nonetheless, our results indicate that French-English bilingual speakers residing in an Anglo-dominant environment are manifesting a bidirectional VOT pattern in the production of voiced stops. In other words, rather than keeping two completely separate language systems, an interaction is occurring between their English and French with regard to voiced (but not voiceless) stops. Namely, their French voiced stops are being coloured by their English, while their French is exerting an influence on their English /b/.

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A NEW RESEARCH ENVIRONMENT FOR SPEECH TESTING USING HEARING-DEVICE PROCESSING ALGORITHMS

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1 Introduction

A common complaint among subjects with hearing loss is the difficulty of understanding speech in the presence of background noise. Speech testing is an integral part of a clinical assessment of hearing loss, complementing the pure tone audiogram and providing useful information regarding speech tolerance and recognition ability as well as helpful cues for hearing-aid fitting [1]. Among other indicators, speech tests are used to determine the speech reception threshold (SRT) defined as the lowest level at which speech can be correctly identified at least 50 percent of the time. The use of such tests is very popular in research as it allows an experimental manipulation of listening conditions in order to study their effects on speech perception with respect to a baseline condition.

In this work, we present a new software-based speech testing environment for research in speech perception. The system is designed to permit the measurement of SRTs under a wide range of listening and processing conditions. This paper briefly describes the conception of the program, outlining its useful features, important design components, and requirements, as well as possible applications.

2 Conceptual design

2.1 Overview

The system is designed to run a speech test using an *adaptive up-down* testing procedure such as the Hearing In Noise Test (HINT) [2]. The program's main graphical user interface provides a set of parameters to specify a listener's hearing profile, listening and processing conditions, and the controls needed to run the test (Figure 1). The software is implemented in Matlab and offers the ability to process sounds through external signal-processing routines prior to presentation to the listener. This functionality is provided by the hearing-device simulator described in Section 2.3. The system's architecture includes a number of Excel files that store important settings, calibration values, hearing-device and test parameters which are easy to modify from within or outside the application. Similarly, test results for a given experiment are exported into an excel spreadsheet to permit analysis as needed.

2.2 Feature set

The program has been conceived in order to permit testing with a range of experimental conditions. Its implementation is not limited to the HINT, which allows future extensions to include additional speech or noise material, languages, and processing conditions. To date, the application includes the following features:

- Testing in quiet as well as in continuous or intermittent speech-shaped noise.
- Testing with the American English or French Canadian HINT sentences (requires permission to use the HINT practice and test sentence lists).
- Testing in monaural or binaural listening conditions.
- Ability to specify arbitrary speech and masker locations (including two preset diffuse-masking specifications).
- A set of simulated programmable hearing devices with the ability to define new ones.
- Three testing paradigms: (1) fixed speech level with an adaptive masker level, (2) fixed masker level with an adaptive speech level, and (3) fixed speech and masker levels with an adaptive distortion threshold.
- Ability to define multiple experiments with independent testing conditions and populations.
- Ability to calibrate the system's sound output using a dedicated Calibration screen.

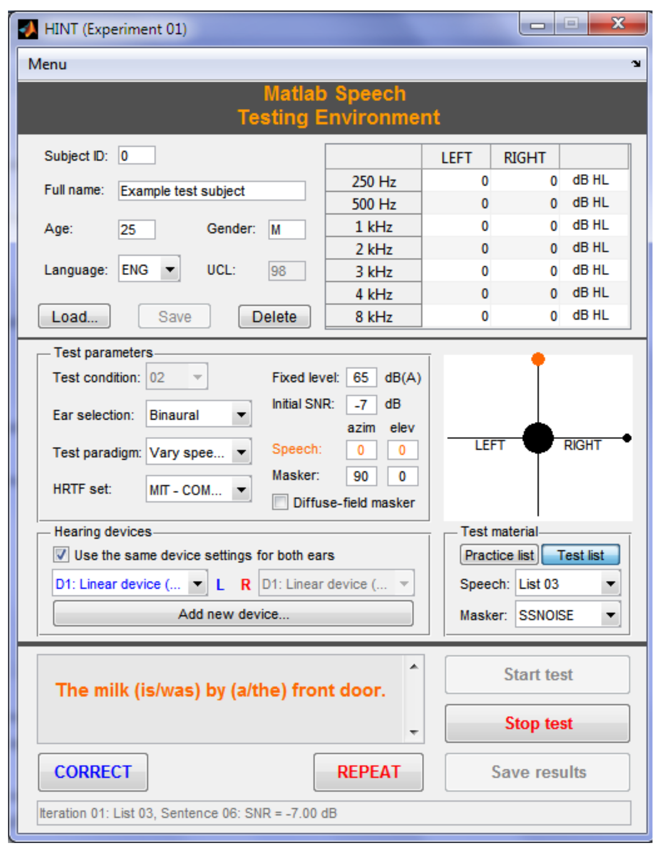

Figure 1: User interface of the Matlab speech testing environment.

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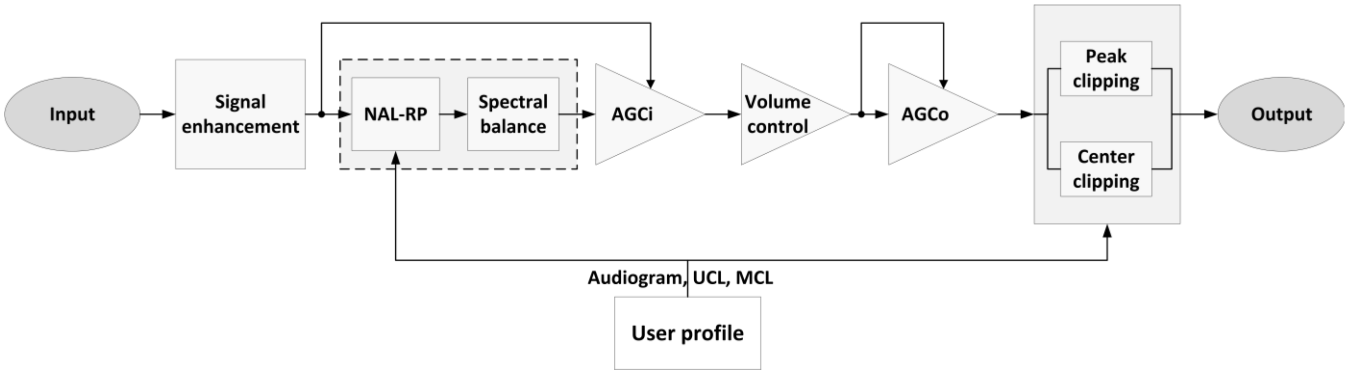

Figure 2: Schematic diagram of the hearing-device simulator

2.3 The hearing-device simulator

A software simulator, implemented in Matlab, provides a set of typical functions commonly performed in real hearing devices. Figure 2 shows the architecture of the hearing-device simulator, which includes the following functions:

- A signal-enhancement module that includes a variety of single-channel noise-reduction algorithms. A complete list of algorithms included to date can be found in [3].
- The NAL-RP linear hearing-loss prescription [4] based on a subject's audiogram.
- A spectral balance function to emphasize low or high frequencies using a spectral tilt with different slopes.
- An amplitude compression algorithm implementing an input-controlled automatic gain control (AGC) system.
- A volume control to specify an overall amplification or attenuation in dB.
- An amplitude compression algorithm implementing an output-controlled AGC system.
- A selection of two algorithms that perform symmetric peak or center clipping.

The hearing-device simulator has been packaged as a standalone toolbox. The Matlab speech testing environment provides a graphical interface to access the simulator and create new devices.

2.4 Spatial listening simulation

The Matlab speech testing environment has been initially designed for testing under headphone listening conditions. The system makes use of a set of head-related transfer functions (HRTFs) to simulate different spatial listening configurations. Two sets of HRTFs (full and compact) from the MIT media lab [5] are included. Additional sets may be integrated into the system if the proper Matlab routines are provided to read the HRTFs. It is possible to perform testing in the sound-field by omitting the HRTFs, and using a set of speakers to achieve the desired spatial configuration.

2.5 System requirements

The software is written for Matlab R2012b for Microsoft Windows systems. Earlier releases of Matlab do not include important functions used in the code (*e.g.* audioread.m). Moreover, the signal processing toolbox is required for the overall system, while some of the algorithms included in the signal-enhancement module require access to the Wavelet toolbox. Finally, in order to carry out speech testing, it is assumed that access to an audiometric room is available along with standard audio equipment to perform calibration.

3 Applications

The speech testing environment presented in this work is intended for use in research environments where Matlab is frequently used as a prototyping platform in the early stages of the development of signal-processing algorithms. It has been employed in a recent study using the HINT speech material to measure subjective SRTs under various spatial configurations and different processing conditions typically encountered with hearing devices [3]. Future extensions of the system being considered include additional test material, such as the Test de Mots dans le Bruit [6].

Acknowledgments

This work was supported in part by the Natural Sciences and Engineering Research Council of Canada (NSERC) and the Queen Elizabeth II Graduate Scholarship in Science and Technology (QEIIGSST).

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MEASURING RHYTHM IN DIALECTS OF NEW BRUNSWICK FRENCH: IS THERE A ROLE FOR INTENSITY?

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1 Introduction

While dialect differences are often associated with lexical items and with the pronunciations of certain vowels and consonants, rhythmic features can also characterize regional varieties of a language. In recent years, researchers have developed metrics that describe rhythmic structure by measuring the variability in the durations of vocalic and consonantal intervals. These rhythm metrics have been used to classify languages into syllable-timed and stress-timed groups, to study rhythm in second language phonology and to classify speakers with dysarthrias. They have also been used to distinguish among the dialects of one language.

However, researchers have noted that these measures have a number of shortcomings. One limitation is the fact that they are based only on duration, to the exclusion of other cues of prominence such as intensity, fundamental frequency and vowel quality. This criticism, made by [1] among others, suggests that an adequate account of rhythm may require a more complex or multidimensional approach that takes into account several prosodic features.

Only a few studies of rhythm have included intensity. Among these, two have examined differences among regional and postcolonial dialects of British English [2,3]. Another study has shown that intensity-based metrics can distinguish between English and Mandarin [4]. Although each study presents different ways of measuring intensity and different formulations of intensity-based rhythm metrics, these studies show that intensity adds significant information towards distinguishing between varieties, going beyond what is captured with duration.

The goal of the present study is to consider the contribution of intensity to the description of cross-dialect rhythmic differences. Specifically we examine how well certain duration-based and intensity-based rhythm metrics can distinguish among dialects of French spoken in New Brunswick (Canada). This is a followup to our preliminary work [5] that shows that regional varieties are a significant source of variation in duration-based metrics.

2 New Brunswick French

New Brunswick is Canada's only officially bilingual province. Francophones, who represent about one-third of the total population of 750,000, live mainly in the northern and eastern regions of the province. Acadian French is the main variety of French spoken. There are three regional dialects: NorthWest, NorthEast and SouthEast, which includes the urban area of Moncton-Dieppe.

Among the features noted in a phonetic study of the SouthEast dialect is the "uneven" (*haché, heurté*) character of the rhythm [6]. Qualitative observations suggest that this impression is due to variable vowel durations and to a strong articulatory force on consonants that occur in the onsets of certain stressed syllables.

3 Methodology

3.1 Speech materials

Speech materials are from the RACAD (*Reconnaissance automatique du français acadien*) Speech Corpus, used for research on the automatic speech recognition of regional varieties of French spoken in New Brunswick [7]. The corpus consists of recordings by 140 native speakers of French from the three regional dialects.

The speakers are stratified by age and gender. There are two age groups: younger adults (average age: 21.1 years) and older adults (average age: 48.3 years). Although the number of speakers in the three dialect groups is uneven, each dialect has a fairly large representation in the corpus: NorthWest (N=26), NorthEast (N=65), SouthEast (N=49). For the present study, we analyzed two sentences that were read by all 140 speakers. Together, the sentences contain about 115 segments.

3.2 Procedure and measurements

Sentences were segmented manually into vocalic and consonantal intervals using *Praat* and following generally accepted segmentation criteria. Almost 14,000 intervals (6,670 vocalic and 7,100 consonantal) were identified.

Durations (measured in msec) were extracted from the segmentation using a script. A *Praat* script was also used to measure the intensity of each vocalic and consonantal interval. We chose the "dB method" which measures (in dB) the mean of the intensity curve of the interval. Because we are not conducting a study of the perceptual basis of the prominence that is rhythm, we limit our focus to this acoustic measure.

We calculated four "local" rhythm metrics, as developed by Grabe and colleagues [8]. These measures are sensitive to sequential contrasts in the speech chain. They focus on differences between immediately consecutive intervals and

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average those differences over a longer unit such as the utterance. The pairwise variability index (PVI) is calculated as the mean of the durational differences between successive vocalic or consonantal intervals. For vowels the index is normalized for speech tempo, (nPVI-Vdur) ; for consonants the raw (non-normalized) index (rPVI-Vdur) is used.

To calculate rhythm indices for intensity, we used the same formulas as for duration. Thus, the two intensity-based metrics studied are nPVI-Vint, for vocalic interval intensities, and rPVI-Cint, for consonantal interval intensities.

4 Results

Descriptive statistics for duration and intensity measures are given in Table 1. The average nPVI-Vdur in New Brunswick French (48.4) is similar to values reported by [9] for other varieties of French spoken in Canada: Hearst ON (43.9), Québec City QC (44.5), Windsor ON (45.5). As a further point of comparison, we note a value of 43.5 for European French, given by [8].

	NorthWest	NorthEast	SouthEast	Dialect
nPVI-Vdur	49.12 (5.3)	47.60 (5.4)	48.90 (5.8)	ns
rPVI-Cdur	61.9 (10.9)	54.48 (8.6)	54.85 (10)	sign
nPVI-Vint	7.65 (3.1)	7.37 (3.0)	5.95 (2.4)	sign
rPVI-Cint	36.30 (4.6)	32.8 (5.0)	34.9 (4.6)	sign

Table 1: Means and standard deviations for four duration- and intensity-based rhythm metrics for the three regional dialects. Statistical significance for dialect is shown in the final column.

Statistical ANOVA tests on rhythmic measures, with dialect as the independent variable, yield differences on one of the duration-based measures: rPVI-Cdur. Post hoc tests show that consonantal duration variability is significantly higher in the NorthWest than in the other two dialects.

ANOVAs show dialect differences for both intensity-based measures. Post hoc tests identify two patterns: NorthWest vs. SouthEast (for nPVI-Vint), and NorthWest vs. NorthEast (for rPVI-Cint).

To observe the performance of these metrics in discriminating among the three dialects, we carried out three linear discriminant analyses. Classifications were done with the duration-based metrics, with the intensity-based metrics and with both types of measures. Results (in Table 2) show that these metrics achieve a modest degree of success in classifying speakers from the three dialects. While intensity-based metrics perform slightly better than duration metrics, the best results are obtained with a combination of both types of metrics.

Type of metrics used	Metrics selected	% correct classification
Duration	rPVI-Cdur	41.4%
Intensity	nPVI-Vint, rPVI-Cint	45.7%
Both duration and intensity	rPVI-Cdur, nPVI-Vint, rPVI-Cint	47.1%

Table 2: Correct classification of 140 speakers from three regional dialects, based on linear discriminant analyses.

5 Discussion and Conclusion

Overall the results show that both duration- and intensity-based rhythm metrics play a role in distinguishing among the three dialects of New Brunswick French. In the three discriminant analyses, the classification results were clearly better than chance. The result that intensity measures achieve a higher rate of classification than duration measures parallels earlier findings in research on the classification of regional dialects of British English [2].

This significant role played by intensity suggests that intensity is an acoustic cue of prominence in New Brunswick dialects of French. While descriptions of European varieties of French note that duration and fundamental frequency are the main correlates of stress, no experimental work (to our knowledge) has studied the acoustic correlates of stress in Canadian varieties. The results of the analyses presented here point to both vocalic and consonantal intensities as likely components of stress.

The broader implication of this study is that it lends support to a multidimensional view whereby different prosodic features contribute to a model of speech rhythm.

Acknowledgments

The authors gratefully acknowledge support from the Social Sciences and Humanities Research Council of Canada.

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USE OF AUDITORY STEADY-STATE RESPONSES IN MEASURING THE ATTENUATION OF HEARING PROTECTION DEVICES

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1 Introduction and objectives

Worldwide hearing loss estimates increased from 120 million people in 1995 [1] to 250 million in 2004 [2]. A common solution to protect workers from noise exposure consists of using hearing protection devices (HPD). Unfortunately, it is difficult to provide hearing protection using an attenuation level that is appropriate for an individual's work environment since present methods of attenuation measurement have limitations [3]. Objective measurements such as field microphone in real ear (F-MIRE) do not assess bone conducted sound. Psychophysical measurements such as real ear attenuation at threshold (REAT) are biased due to the low frequency masking effects from test subjects' physiological noise and contain variability of measurement due to subjective responses. The present study reports an attempt to overcome the limitations of these methods through the recording of auditory steady-state responses (ASSR). Due to the time consuming nature of ASSR recording, the study was conducted using only two stimuli having 0.5 and 1 kHz carriers.

2 Methodology

Ten volunteers (8 males – 2 females) with ages from 20 to 29 and thresholds below 20 dB SPL (from 125 Hz to 8 kHz) were assessed. A typical experimental procedure included four steps:

2.1 Step 1: Custom earplugs molding

Each experiment started with the molding of the custom earplugs used throughout the recordings for each subject. In selecting the earplugs, we were sensitive to the fact that, when a subject is wearing earplugs, the stimulation levels must be adjusted to ensure that the sound pressure levels in a subject's ear canals are approximately the same in both open and occluded conditions. Accordingly, attenuation at 500 Hz and 1 kHz should be moderate (e.g. 10-20 dB) and stimulation levels should not exceed about 75 dB SPL. This ensured, in case of bad fit, that a subject would not be exposed to an excessive noise level. Lastly, due to the somewhat long duration of the experiment (e.g. 90 minutes), the earplugs should be comfortable.

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Sonomax Self-Fit™ earplugs are suitable for this experiment. These are fitted to the user's ear by injecting medical-grade silicon between a rigid core and an expandable membrane. The earplugs are designed to include an inner bore of constant length and diameter that permits the temporary insertion of a microphone probe and the permanent insertion of a passive acoustical filter. We used, a brown-colored filter, in the set of filters provided by Sonomax, which provides an attenuation of 10 dB at 500Hz and 15 dB at 1 kHz.

2.2 Step 2: REAT measurements

The REAT method is commonly considered as the "gold standard" and is included in many worldwide standards. REAT calculates HPD attenuation as the difference between open-ear and occluded-ear hearing thresholds for human subjects. REAT measurements were conducted according to ISO 4869-1:1990 standards by using the LabVIEW™ based "REAT MASTER™" Nelson Acoustics software which is an automatic Bekesy audiometer program designed to provide a stimulus and track responses for sound field hearing threshold determination in both normal and occluded conditions. Stimuli used to conduct REAT measurements were warble tones at 0.5 and 1 kHz computer generated and presented via a Sennheiser headphone.

2.3 Step 3: F-MIRE measurements

The F-MIRE method calculates the noise reduction (NR) as the difference between the residual SPL in the ear canal and the external SPL, measured simultaneously by two miniature microphones positioned inside and outside the ear-plug. F-MIRE measurements were conducted by using "SONOPASS™" software which measures the NR and then applies compensation factors to provide an effective individual attenuation rating. The stimulus used to conduct F-MIRE measurements was a computer generated pink noise that was presented at 85 dB SPL via a speaker in frontal incidence.

When using Sonomax Self-Fit™ Hearing Protectors, sound can travel to the middle ear along two paths: a solid path and an air path. The first path corresponds to the sound going through the HPD material itself and the second path corresponds to the sound going through the filter. During F-MIRE measurements, the microphone is plugged into the hole for the filter and the HPD is "full-blocked". The measured attenuation reflects the reduction of sound through the HPD. If the filter is not a full-block filter (as with brown filter), the software decreases the attenuation value to match that of the filter. For that reason, F-MIRE results should be the same for each individual. Since the F-MIRE method cannot measure the real attenuation of the ear plug with a filter, the F-MIRE measurements realized in this study have been used to verify the conformity of the custom HPD and to provide an initial prediction of the REAT.

2.4 Step 4: ASSR recordings

ASSRs are electrophysiological responses [4], recorded from the human scalp, evoked by one or more carrier frequencies (F_c) that are each amplitude-modulated at a specific frequency (F_m). In practice, if the subject hears such a stimulus, a peak will appears in the EEG frequency spectrum of the subject at F_m. Since amplitudes and phases of ASSR are quite well correlated with the intensity of stimulation, it may be possible to measure earplug-attenuation by recording ASSRs using both normal and occluded conditions. ASSR recordings and stimuli generations were conducted by using the LabVIEW™ based "MASTER SYSTEM™" Rotman Research software. Stimulus characteristics are reported in Table 1.

3 Results and discussion

ASSR-based "physiological" attenuation has been calculated as the average difference between the normal and occluded conditions using linear least-square regression of ASSR amplitude and phase data. As seen on Figure 1, the amplitude-based estimate of attenuation computed with the highest stimulation levels (A_1) is substantially the same as the one computed with the lowest stimulation level (A_2). This attenuation effect was not as consistent in the phase data (lower graphs of Figure 1). This finding suggests that physiological attenuation seems to be more accurate when calculated from the amplitude rather than phase of the ASSR.

Physiological attenuation estimates were expected to be different from REAT-based attenuation results, because we used supra-threshold stimulation levels to deter any low-frequency masking effect. However, Wilcoxon tests failed to show any statistical difference between REAT and ASSR results (p value > 0.1). This suggests that the effect of low-frequency masking may not be as large an influence as previously assumed.

Although other electrophysiological methods have been adapted in the past for measuring the attenuation of HPD [5], no study has considered using ASSRs. The present study seeks to ascertain whether it is possible to objectively measure the attenuation of HPD using ASSRs collected in the same subject both with and without protectors. The results are encouraging: we successfully measured the attenuation in every volunteer who participated but further research, using an extended frequency range, should be done to explore this hypothesis.

Stimulus	F_c	F_m	AM%	Open	Occluded
#1	500 Hz	40 Hz	100 %	45 to 65 dB (10 dB step)	55 to 75 dB (10 dB step)
#2	1 kHz	41 Hz			

Table 1: Characteristics of the amplitude-modulated tones used in the ASSR experiment, for both normal and occluded condition.

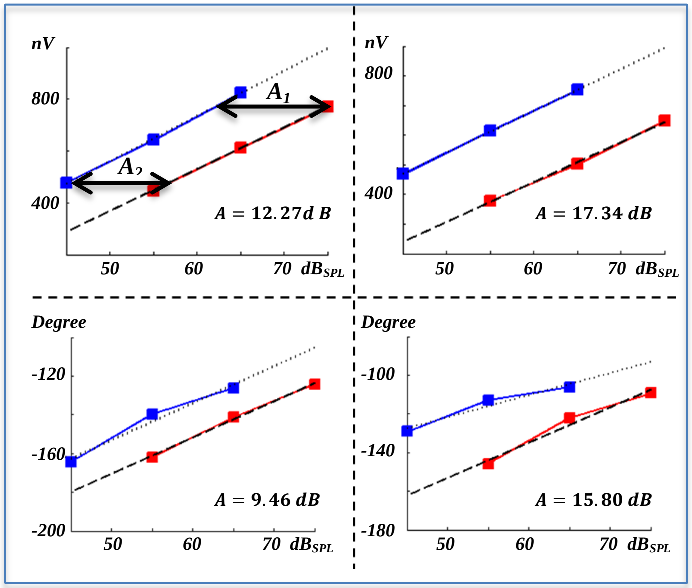

Figure 1: Grand mean ASSR amplitude (upper plots) and phase (lower plots) as a function of stimulation intensity, at 0.5 (left plots) and 1 kHz (right plots). Occluded results are represented by the red curves and normal results, by the blue curves. The linearity of the responses is represented by the black dotted lines. The «physiological» attenuations have been calculated as the mean difference between the two curves (A= 0.5*[A1+A2)]).

Acknowledgments

The authors wish to express their appreciation to SONOMAX HEARING HEALTHCARE INC. (Montréal, Canada) for providing the custom earplugs and to NSERC (Natural Sciences and Engineering Research Council of Canada) for their support to this project.

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VALIDATION OF A FINITE ELEMENT CONTINUUM MODEL OF VOCAL FOLD VIBRATION

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1 Introduction

The accurate simulation of vocal fold vibration is contingent on appropriate formulation of the system's equation of motion. Lumped element and continuum formulations have been applied in previous studies to characterize this vibration under both normal and pathological conditions. The present study seeks to validate an in-house finite element code of vocal fold vibration, and to examine the resultant effects of an added unilateral sessile polyp on the fundamental frequency of vocal fold vibration. Validation is performed with a direct comparison of natural frequencies and modes of a numerical study by Luo et al. [1]. The development of the pathological vocal fold model is subsequently discussed, and trends are presented.

2 Methods

The finite element method (FEM) is applied to a discretized vocal fold system to formulate the general eigensystem given by

$$[K]\phi = \omega^2[M]\phi \qquad (1)$$

where the mass matrix, $[M]$, and stiffness matrix, $[K]$, are developed using the FEM. For an N degree of freedom system, equation (1) can be solved for N pairs of eigenvectors, ϕ, and eigenvalues, ω^2 [2, p. 786]. Geometries are discretized using linear strain elements, and a combination of analytical and numerical techniques are used to evaluate the mass and stiffness contributions of each element.

3 Model Validation

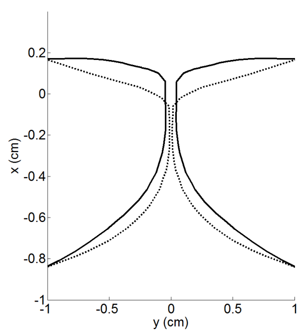
A multi-layered, three-dimensional continuum model of a healthy vocal fold is presented in [1]. This model documents the material properties, boundary conditions, and geometry of each layer in detail [1, p. 9330, 9316]. Luo et al. adopt the immersed boundary method to solve for the first four natural frequencies and mode shapes of the model, which are used in the present study for direct comparison. Convergence of these natural frequencies was observed in the FEM model for increasingly fine meshes. The results obtained from the finest mesh are used for comparison.

Table 1 contains a comparison of the first four computed natural frequencies with those reported in [1]. The maximum computed percent difference is 5 %, which corresponds with the fourth mode of vibration. Since a comparison is being drawn between two numerical studies, exact agreement would not be expected. Nonetheless, the agreement of the FEM model with the solution from [1] was deemed similar enough to conclude the computed solutions are accurate.

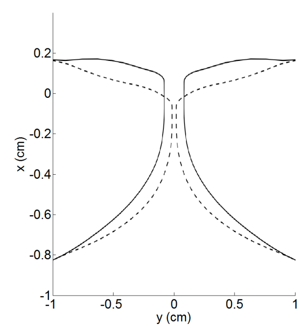
Table 1: Comparison of computed natural frequencies of the healthy vocal fold model

Mode number	Luo et al. (Hz)	FEM model (Hz)	Percent difference (%)
1	114	110.1	3.5
2	125	120.5	3.7
3	133	128.4	3.5
4	144	136.9	5.0

Further comparison is drawn qualitatively between the computed mode shapes. For brevity, only a comparison of the first mode is reported, in figure 1. This figure illustrates the outer periphery of the coronal cross-section, midway between the anterior and posterior vocal fold surfaces. Lateral and vertical displacements of the positive and negative eigenvectors appear to be in agreement.



(a) Computed mode shape using the FEM



(b) Computed mode shape from [1]

Figure 1: Comparison of the first mode of vocal fold vibration. Pictured is the coronal cross-section.

4 Development of a Pathological Model

Polyps are unilateral lesions which form on the cover of a vocal fold. Sessile polyps have a balloon-like appearance, and manifest as half-ellipses which protrude from the medial surface of the vocal fold [3, p. 268]. Sessile polyps have been observed between 0.3 mm to 0.7 mm in length [4, p. 129]. The material properties of polyps have a wide range of reported values, though typically, the polyp presents increased stiffness relative to the surrounding tissue.

Based on this data, a model of a vocal fold afflicted with a sessile polyp was developed, with its geometry presented in

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figure 2. Vocal fold geometry and orthotropic material properties are based on Luo et al, [1, p.9330]. The polyp is represented as a sphere whose center and diameter are treated as variable parameters. The polyp's center, c, is situated midway along the medial surface in the y direction, and may vary along the anterior-posterior direction, z, between 0.3 mm $\leq c \leq 13.7$ mm. The length, d, is bound between 0.3 mm $\leq d \leq 0.7$ mm. The polyp is treated as an isotropic material, with a fivefold increase in stiffness relative to the transverse Young's modulus of the ligament, a density of 1.1 g/cm^2, and a Poisson ratio of 0.3. Anterior, posterior, and lateral surfaces of the vocal fold are held fixed, while remaining nodes are free.

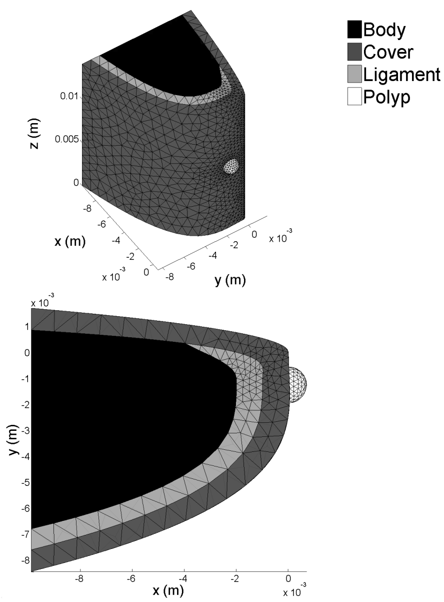

Figure 2: Pathological vocal fold mesh. The polyp position may vary along the anterior-posterior direction, z, and the polyp radius may vary from 0.3 mm to 0.7 mm.

The effects of varying c and d on the fundamental frequency are assed with two sets of simulations, similar to a study by [5]. The first set of simulations varies 0.7 mm $\leq c \leq 7$ mm at $d = 0.7$ mm for 10 trials. The second set of simulations varies 0.3 mm $\leq d \leq 0.7$ mm at $c = 7$ mm for 5 trials.

5 Results and Discussion

The effect of varying the position of the polyp along the anterior-posterior direction is shown in figure 3. As the polyp moves towards the center of the vocal fold ($c = 7$ mm), the fundamental frequency reaches a minimum. Since the vocal fold geometry is symmetric, the same trend would be expected for a set of simulations which varies 7 mm $\leq c \leq 14$ mm. For this dataset, a linear trend line was fitted. This varies from [5], which fit a quadratic curve, but otherwise, both studies present the same trend between frequency and position. Variance may be due to the smaller dataset used in [5].

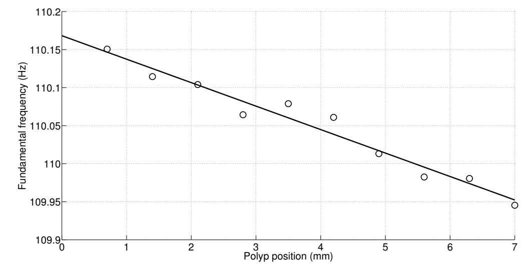

Figure 3: Effect of polyp position on fundamental frequency

The effect of varying the size of the polyp is shown in figure 4. Asymptotic behaviour is observed at the upper and lower bounds of d. Smaller polyps will have negligible influence on frequency, and inversely, larger polyps have a profound influence on frequency. This trend corresponds with studies which examine the damping of the mucosal wave on the vocal fold surface due to localized stiffness increases of the polyp.

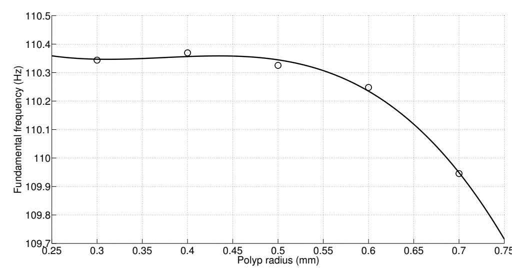

Figure 4: Effect of polyp size on fundamental frequency

The validation process of the FEM code is integral for future uses as a predictive tool. Future studies will examine mode shape variation under the influence of polyps, as well as acoustic sound radiation in a time domain analysis.

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USING DISCRETE COSINE TRANSFORMS TO CHARACTERIZE TONES IN TWO ATHABASKAN LANGUAGES

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1 Tone in the Dene languages

Tone in North American languages has received very little phonetic attention. This paper is a preliminary analysis of the production of lexical tone in two related bitonal (H/L) Athabaskan languages spoken in Northwestern Canada: Dene Sųłine (Chipewayan) and Tłįchǫ Yatiì (Dogrib). The Dene languages are divided into two groups, those without tone (Alaska, Yukon and the Pacific coast) and the tone languages of the inland Dene and the Apachean languages. Tonogenesis arises from the incorporation of stem final glottalic elements into the stem nucleus, through an intricate set of reconstructed stem alternation patterns [1], [2]. The most commonly reported phenomenon is a type of tonal polarity, which reoccurs in daughter languages [3]. Tłįchǫ Yatiì (TY) and Dene Sųłine (DS) are examples of two-tone contrasts with opposing polar-type tone systems. In this study we looked at 2-syllable words taken from existing recordings of word lists by native fluent speakers in Tłįchǫ Yatiì ('H marked') and Dene Sųłine ('L marked'). There were 3 speakers for each language. Given the default status of the opposite tone in each language, we are investigating differences in the realization of the tones between the languages. If tonal implementation is affected by morphological, lexical or grammatical factors, we expect differences due to the asymmetries caused by the distribution of the default tone.

2 Discrete Cosine Tranform

We use the discrete cosine transform (DCT) to characterize tone trajectories of words in these two languages. The DCT is a transformation that decomposes a spline into a set of coefficients (k_0-k_n) from which the spline can be reconstructed; much like a Fourier transformation. Watson and Harrington [4] showed that k_0 is proportional to mean f_0; k_1 to slope and k_2 to (parabolic) curve. The DCT coefficients are all real numbers which provide numerical correlates for trajectory shape. The studies in [4] and [5] used these coefficients very successfully to characterize formant tracks. In this paper we use them to characterize pitch tracks in tone.

3 Study

3.1 Procedures

Two syllable words were extracted from recordings of word lists made by 6 native speakers (3 DS and 3 TY speakers). For Tłįchǫ Yatiì, which has phonological length, only words with short vowels were chosen. In the recordings each subject had repeated the words three times. For DS all three speakers produced all 9 of the words given in Table 1a and all repetitions were used. For TY not all speakers produced all the words and the distribution is shown in Table 1b. As well, in TY speaker P consistently produced noticeable list intonation so only the first two tokens of her examples were used. Speaker L also produced list intonation for gobò and kwìgha so only the first two tokens for those two items were used. All used tokens were annotated in PRAAT [6] and the vowels segmented out. Eleven equally spaced F_0 measurements were taken from each vowel, the values were converted to Barks and the DCT coefficients for each contour were calculated using the emu package [7] in R [8].

3.2 Results

Figures 1 and 2 show the interactions of k_0 (mean) and k_1 (slope) for DS and TY respectively. The plotting includes a division by syllable: the number indicates either the first or second syllable (1 or 2) and the tone is indicated as H or L. Tone is also indicated by colour. A plot along the central, vertical line (k_1 = 0) indicates a flat contour, plots to the right (positive k_1 values) correspond to a negative slope or a

a) **Dene Sułine**

WORD	TONE	GLOSS	SPKRS
yéłk'éth	HH	*he shot it*	S A H
yédzi	HL	*he caught it*	S A H
ghedéł	LH	*they are walking*	S A H
thitth'í	LH	*I pinched him*	S A H
destur	LL	*I mix it*	S A H
ghegał	LL	*he is walking*	S A H
ghiził	LL	*I screamed*	S A H
hegheth	LL	*he is itchy*	S A H
hestį	LL	*I succeeded*	S A H

b) **Tłįchǫ Yatiì**

WORD	TONE	GLOSS	SPKRS
degho	HH	*bark*	M L
gokwi	HH	*axe*	M L
sechi	HH	*younger brother*	M L
ekwò	HL	*caribou*	M P
gobò	HL	*abdomen*	P L
gokè	HL	*foot*	M P L
gokwì	HL	*head*	M P L
dìga	LH	*wolf*	M P
kìwgha	LH	*hair*	M P L

Table 1: The words used for each language and the speakers for each word.

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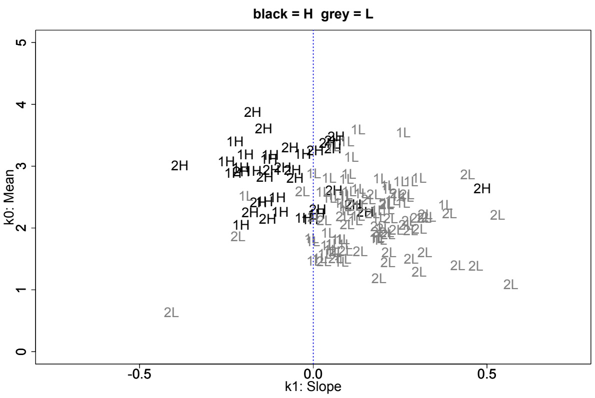

Figure 1: Interactions of k_1 (mean) and k_2 (slope) for high and low tones in Dene Sųłine.

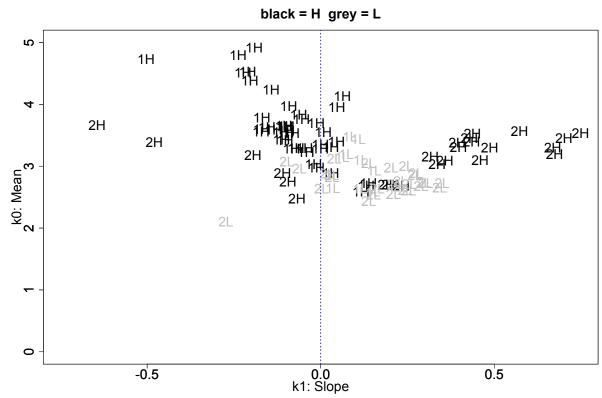

Figure 2: Interactions of k_1 (mean) and k_2 (slope) for high and low tones in Tłįchǫ Yatiì.

falling contour with the slope increasing as the k_1 value increases. Negative k_1 values correspond to a rising slope and increasing pitch contour. The mean corresponds directly with k_0 so a higher k_0 equals a higher pitch.

The calculations for each language were fitted separately to a linear model with k_0 (pitch) as the dependent variable and tone as the independent variable (high tone as the intercept). Results were significant for both DS ($F(1, 142) = 47.59$, $p = 1.60 \times 10^{-10}$) and TY ($F(1, 101) = 49.09$, $p = 2.79 \times 10^{-10}$). Similar models were fitted for k_1 (slope) but only DS showed significance ($F(1, 142) = 77.88$, $p = 3.654 \times 10^{-15}$).

4 Discussion

Visual examination of the graphs shows some interesting clustering for both languages. DS (Figure 1) shows fairly clear grouping by tone. The H tones appear to group to the left of the 0 slope line suggesting a slightly rising contour while the L tones show the opposite tendency.

Statistical analysis of the DCT coefficients for both languages show a significant difference in k_0 (pitch) between H and L tones. The languages differ when it comes to k_1 (slope). DS shows a significant difference between the tones in this regard. A closer look at the results also shows that H tones in DS do not differ significantly from the intercept (or 0) which would correspond to a level contour. This suggests that speakers of DS employ at least 2 different phonetic features to differentiate tone.

The graph for TY (Figure 2) shows binary clustering of the H tones with most of the 2H syllables showing a high k_1 well removed from the 1H syllables. This suggests word-final falling contours which may be a boundary tone effect. There was no significant difference between the k_1 (slope) values by tone suggesting that slope does not play a role in tonal differentiation in this language.

5 Conclusion

The findings suggest that the realization of the tones is quite similar between the two languages, with some interesting differences. There are differences in the realization of H tones in stems in Tłįchǫ Yatiì that may reflect an interaction of tone and vowel length [9]. Also, in Dene Sųłine, H tone tends to resist a fall in the stem (final syllable), unlike its L tones and both tones in Tłįchǫ Yatiì. This study implicates the importance of the differences in tonal specifications and alignment among the Dene tone languages in understanding tone patterns and tonogenesis. Future studies involve the interaction of tone, especially H tone and tonal alignment with vowel length, and the distinction between the realization of tone in the stem versus pre-stem domain.

Acknowledgments

This project has been made possible due to funding provided by the Volkswagen Foundation and the NSF. Neither is responsible for the opinions or analysis herein.

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AUDITORY CATEGORIES AND LARYNGOSCOPIC/ULTRASOUND IMAGES IN THE *iPA Phonetics* APP

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1 The sounds of the IPA in an App

iPA Phonetics is a self-contained iOS application created at the University of Victoria with the intention of illustrating the auditory categories and acoustic sound distinctions of the International Phonetic Alphabet (IPA) as well as the articulations that correspond to the production of those sounds. The App illustrates IPA consonant and vowel sound production with high-quality colour videos and illustrates an inventory of phonetic voice qualities with audio files whose labels are mapped onto a graphic that orients the categories geographically on the oral and laryngeal parts of the vocal tract. The format of the consonant chart in the App is an expanded elaboration of the official IPA chart, including many more characters and sound combinations, all of which are implied in phonetic theory. The vowel chart is also in IPA format, with one added symbol. The App gives users of Apple iOS mobile electronic devices (iPhone, iPad) the ability to hear, view, manipulate, and compare the sounds and articulations symbolized in the charts and voice quality table. For phonetic training, the App also has a ‘matching game’ function, which presents users with random video files to test their knowledge of the phonetic symbols and sounds. The videos are endoscopically filmed views of the oral vocal tract and of the laryngeal vocal tract. Laryngeally articulated sounds are also accompanied by a choice of three ultrasound views of the laryngeal vocal tract.

2 Purposes and functionality of the App

The purpose of this free App [1] is to introduce the general public as well as specialized users of IPA symbolization, via iPad/iPhone technology, to the auditory/acoustic inventory of possible speech sounds of the languages of the world and to how each sound is physically articulated. The App is a learning tool created in a university environment for the benefit of students, teachers, researchers, and the wider community at large. The format of the App is illustrated in Figure 1. By touching a symbol, each sound category can be listened to and viewed in the form of close-up orally-filmed endoscopic videos of the vocal tract. Images/audio may be sped up or slowed to view articulatory detail and gain an understanding of the phonetic classification system and the degrees of auditory distance between categories in the taxonomy. Videos of consonant and vowel categories can be expanded to full screen, with a slider bar for manual control. Both the consonant chart and vowel chart can toggle on/off the IPA Numbers of all symbols [4]; a useful resource for coders. The consonant chart has a zoom in/out feature.

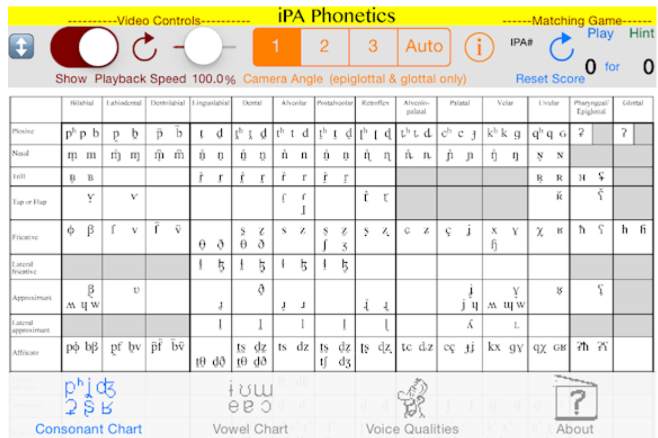

Figure 1: The main page of the *iPA Phonetics* App, showing a portion of the Consonant Chart, option selection bars, and links to the Vowel Chart and Voice Qualities pages and information page.

3 Auditory/Acoustic/Visual properties

3.1 Visual design

The chart format follows the elaborated inventory of speech sounds of the IPA published in the revised *Handbook of Phonetic Sciences* [2]. Categories are added to the standard set of IPA places and manners of articulation drawing on the ExtIPA chart [3,4] for articulations considered ‘normal phonetic’ possibilities as opposed to disordered/clinical, i.e. the Dentolabial and Linguolabial columns. An Alveolo-palatal column is added, expanded from its brief mention in the official IPA chart. Pharyngeal is considered identical in articulatory place to Epiglottal. There are four major divisions in the consonant chart (separated by bold lines): the first three columns are Labial; the second five, Front Lingual; the third four, Back Lingual; and the last two, Laryngeal. Symbols are included for as many cells in the matrix as phonetically practicable, given articulatory possibilities. Shaded areas denote phonetic impossibilities. Diacritic placement follows IPA principles. The Plosive row explicitly includes aspirated stops. The Nasal row is fully elaborated for all places with a voiceless counterpart in each cell. Trills involve the lips, tongue tip, uvula, and the aryepiglottic folds of the Pharyngeal articulator (using the IPA symbols for Epiglottal fricatives). Labiodental/Bilabial flaps, a Uvular tap (controversial in past IPA charts), and a newly designated Pharyngeal tap appear. Fricatives contrast rows for sibilance or grooving. Approximants include a set of ‘approximated fricatives’ [5]. Affricates are elaborated, in parallel to Fricatives, including for Pharyngeals. Implosives, Ejectives (with affricates), and Clicks (the sparse set from the IPA chart rather than a full set [6]) are added as manners in this chart. The vowel chart, in the format of the 2005 IPA chart with one unrounded near-close back symbol added, has four playback options: Long, Short, Pairs, and 'sweeps' along various axes of the chart. This is a novel programming feature for phonetics illustrations, taking advantage of iOS capabilities.

The visual presentation of articulations in this App is distinguished by its use of endoscopic photography. Views of Labial sounds show the outside of the mouth, views of Lingual sounds look into the mouth, and views of Laryngeal sounds are taken with the rigid endoscope over the back of the tongue to see into the throat (larynx). Symbol selection, performance of the phonetic categories, and laryngoscopic filming were carried out by the first author. The design and programming of the App were accomplished by the second author. Simultaneous ultrasound data capture and graphic depiction of ultrasound/larynx views were done by the third author. Ultrasound images of the lower region of the vocal tract, previously never included in a database of articulatory categories, complement the lingual ultrasound images of oral sounds in other databases [7].

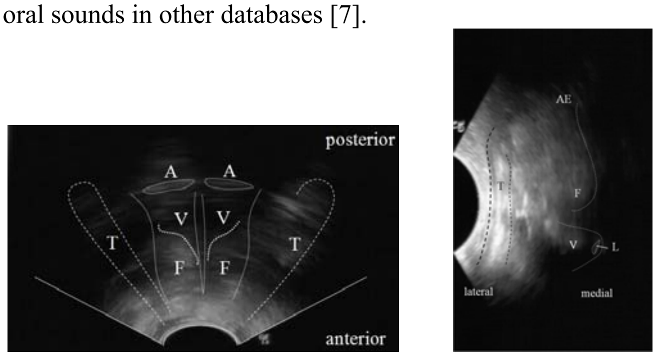

Figure 2: Axial convex (left) and Vertical convex ultrasound (right). A, arytenoid cartilages; V, vocal folds; T, thyroid cartilage; F, ventricular folds; L, vocal ligament; AE, aryepiglottic folds.

The App also has a Voice Qualities page with audio of oral and laryngeal categories, oriented on a static graphic of the vocal tract representing the Laryngeal Articulator Model of speech production [8]. Each category is listed by label and grouped into strategic articulatory regions, especially in the laryngeal vocal tract, as an exploratory feature for users.

3.2 Technical aspects

Video/audio clips for each set of symbols were captured using a KayPENTAX 9100 Rhino-Laryngeal-Stroboscope, 9105 70º-angle rigid oral laryngoscope (hand-held), a 35mm lens to view oral articulations and a 28mm wide-angle lens to view laryngeal articulations, connected via a Panasonic GP-US522 camera to multichannel-synchronizing software and procedures developed by the second author.

Ultrasound images were captured simultaneously with the laryngoscopic images of Laryngeals using a GE portable LOGIQe R5.0.1 system and 8C-RS convex probe to image supraglottal laryngeal involvement (e.g. for Glottal stop) and 12L-RS straight-line probe at a relatively shallow 2-4 cm depth on the neck and about 2-4 cm of the vertical dimension for clear resolution of laryngeal structures to image larynx height changes (e.g. for Pharyngeal/ Epiglottals). This is a novel laryngeal technique that differs from the approach usually taken in oral lingual ultrasound data capture. The three viewing options are an axial convex view (Figure 2), a vertical convex view (Figure 2), and a vertical flat (straight-line) view showing neck muscle movement. Anatomical sections of larynx models are included with corresponding labelled ultrasound images in an 'INFO' page, viewable via the toolbar, to illustrate the detail of what can be seen using laryngeal ultrasound [9].

4 Challenges in producing the resource

There are a number of technical and theoretical challenges in producing a resource of this nature. The first is to select phonetic categories that reflect as exhaustively as possible the auditory/acoustic separation between 'cardinal' speech sound values. The second is to perform these canonical values with accurate articulatory gestures and sound quality, while abating the noise of the laryngoscopy and ultrasound instruments in the audio signal, recalibrating the intensity of the light source and camera resolution, holding the scope (and ultrasound probe) in an optimal position for filming, then post-processing the video to obtain phonetically ideal auditory/acoustic distinctions and uniform video/colour and audio quality across all final files. The virtue of the iOS environment, from the point of view of technical implementation, is that novel approaches to displaying the phonetic subject matter can be seen and tested by a wide crowd of users, who then have a role to play in developing the effectiveness of the educational resource.

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Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Compensatory Vowel Formant Production With Artificial Voicing Source In Response To Real-Time Auditory Perturbation

Takashi Mitsuya, Kevin G Munhall, David W Purcell

Speakers monitor their own voice to maintain and control accuracy of speech production. Studies have shown that when speakers receive auditory feedback that is perturbed in real time, they change the production of the perturbed parameter (e.g., vowel formants) to minimize the difference between the intended versus the heard acoustics (i.e., to minimize the error). It has been assumed that the speech motor control system processes auditory feedback from self-produced utterances for the purpose of estimating error signals. However, little is known about this assessment process for self-productions. Here we examined whether the speech motor control system can produce compensatory changes in production of vowel formants when the vocal source is static and not one's one. In the current study, the first formant was increased in real time auditory feedback while speakers produced a short sentence "Ted said head" with their normal voice (natural voicing) for one group or with an electrolarynx (artificial voicing) for another. The formant structures of the produced vowels were no different across the conditions. More importantly, speakers compensated to perturbations in a similar manner, such that the magnitude of compensation across conditions was not significantly different. These results show that compensatory behavior can be elicited without the acoustic characteristics associated with one's own voicing.

Detection Of Deviations In Straight Formant Transitions

Michael Kiefte

The present study determined just-noticeable-differences (JND) in deviations from linear formant trajectories. Diphthong-like stimuli were manipulated by inserting a point of inflection into the second formant transition for the vowel /ei/. A large number of conditions were tested including rate of change, direction (downward versus upward), total duration, presence versus absence of other formants, average formant frequency, and acoustic source. With the exception of conditions in which the acoustic source was manipulated, all thresholds for deviation from linear formant transition converged at roughly the same absolute frequency (e.g., even when the average formant frequency was raised by 50 Hz). The threshold frequency was significantly different when fundamental frequency was manipulated, when the source was replaced with white noise (i.e., whispered speech), and with sinewave analogues of formants (sinewave speech). This suggests an interaction between formant and fundamental frequency. Results from these experiments along with several hypotheses are presented.

Some Concepts Underlying An Embodied Theory Of Speech Sounds

Bryan Gick

Whether for speech or for any volitional movement, we have the deep-rooted sense that humans are born endowed with defined body parts and must learn to control them. An alternative, embodied, view is that the use and control of a body part is encoded in its physical structure. This view is a logical consequence of the long-standing notion that movement is controlled via functionally defined neurophysiological structures, (sometimes called muscle synergies, coordinative structures, or the like). Under this view, we do not learn to control body parts; rather, we learn to have useful body parts. Of course, we are born endowed with a large set of useful body parts that include – indeed, are defined by – the neural structures that control them (e.g., for breathing, swallowing, locomotion, etc.), and we learn to build new ones. Once we have the right parts, we know – by definition – how to control them. The parts know what to do because they are not mere anatomical structures but neurophysiological ones. Taking this view to speech, an embodied theory of speech sounds – of phonetics and phonology – is one that acknowledges that the communicative actions we perform are determined by the effectors/devices we define, the body structures we habitually use. Thus, we do not control an inventory of speech sounds or movement control parameters; rather, we control an inventory of neurophysiological body parts each of which has been built, either through phylogenetic or ontogenetic encoding via repeated use, to produce a sound or some ecologically discrete aspect of a sound or other communicative event, where each part governs its own degree of freedom in our action space. Some particulars and implications of such a theory of speech are discussed.


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UNDERWATER ACOUSTICS - ACOUSTIQUE SOUS-MARINE

CRICKETS: TEMPERATURE DEPENDENT BACKGROUND SOUND

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1 Introduction

The acoustic environment outdoors is characterized by the sounds of human activity and the sounds of nature. Sound from human activity follows a common pattern [1] that is comparable to patterns in traffic volumes [2, 3]. The typical diurnal pattern in urban areas rises sharply between 0500h and 0700h, maintains slightly reduced levels until approximately 1900h, and then drops off to a minimum level between 0300h and 0400h. In contrast, the sounds of nature, largely driven by weather and animal activity, are more variable. Animal activities also follow patterns, but are more variable with their seasonal life cycles.

The impetus for this evaluation was an unusual pattern in long term sound measurements. The sound levels, which were much higher than expected, and minima which occurred at unusual times of day led to further investigation and identification of the source.

2 Results

2.1 Data Collection

Sound levels reported in this study were measured in rural southern Ontario. The primary location is sheltered by a stand of mature trees, with agricultural fields in the vicinity and a rural road at some distance. The sounds of nature dominate during nighttime, while a large increase in road traffic volume influences daytime sound levels.

Initial data from the primary location was led to comparison to evaluate variation year-to-year, seasonally, and by degree of urbanization. These smaller data sets are only addressed qualitatively here.

Measurements were conducted over several months using Larson Davis 820 and Bruel+Kjaer 2250 sound level meters, each equipped with their respective environmental protection kits. Minimum, maximum, average L_{EQ}, and statistics were recorded on an hourly basis. Manufacturers' specifications governed the validity of data collected. Where weather conditions were outside of meter's specified range, the data was excluded from consideration. Audio data was not available for the extended data set.

Meteorological data including temperature, wind speed, precipitation, and relative humidity were collected at the same time as the sound level measurements. Equipment collecting the meteorological information was located away from the stand of trees. Wind speeds therefore do not reflect the sheltering effect of the tree stand.

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2.2 Sample Data

The pattern of sound levels that initiated the investigation occurs overnight. Daytime hours are not notably different from an urban sound pattern. As shown in Figure 1, there is a sharp rise of approximately 10 dB that falls back to a similar minimum sound level 11 hours later. The pattern was most prominent between 45 and 55 dBA, although a range of 35 to 60 dBA was observed.

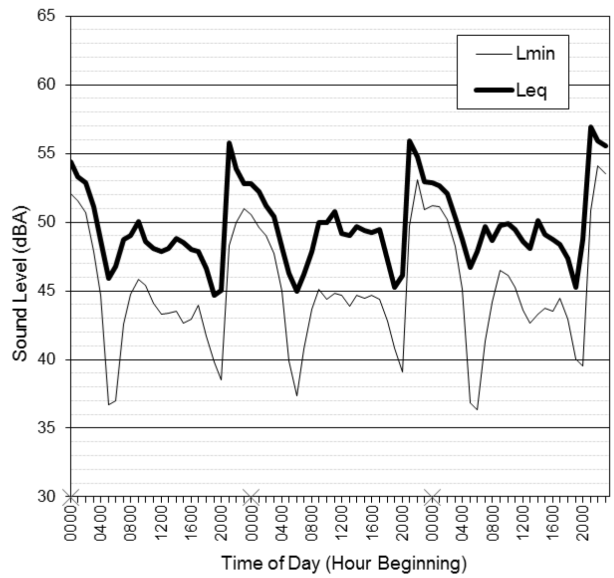

Figure 1: Typical Observed Sound Level Pattern by Time of Day.

Start and end times of the pattern are very consistent. The sharp rise occurs very consistently at 2000h and gradually falls off to a minimum at 0700h. These coincide with sunrise of 0730h and sunset of 1930h during the same period.

The pattern was observed as early as July 25th and as late as September 30th. Review of the statistical sound levels suggested that the source was present for large portions of each hour, and was not a short duration event.

2.3 Correlation with Temperature

In the absence of obvious mechanical sources of noise, correlation was sought with weather parameters: relative humidity, wind speed and temperature. Neither wind speed nor relative humidity showed a relationship with the sound level data. However, there was correlation with temperature.

Daytime sound levels showed no temperature correlation. Sound levels at the beginning and end of the pattern also were not as strongly correlated with temperature. The window beginning 2100h and ending at 0500h on the following morning was therefore used. The resulting correlation between sound level and temperature is show in Figure 2.

Analysis was based on the 400 remaining data points. The data shows a distinct rising trend. Sound levels rise by 5 dB for a temperature increase of 4 °C beginning around 10 °C and 40 dBA.

2.4 Comparison with Other Data Sets

The review of additional data sets looked for the pattern:

- at the same location in other years;
- at the same location in other seasons;
- at other locations in other seasons; and
- at more urban locations in the same season.

Comparison of the primary dataset with the same location at the same time in other years showed that the pattern was also present although the sound level did not always show as much increase with temperature. The pattern was not present in other seasons of the same year. A more wilderness site, measured earlier in the year, also did not show the pattern. Finally comparisons were made with a number of more urban sites. Late summer measurements made in the quieter yards, parks, and unmanaged lands in urban areas had sound levels that were in the upper range of levels observed in the main data set. A correlation between sound level and temperature was not observed.

2.5 Audio Data

A small amount of audio was collected during comparable measurements in a subsequent year. The chirping of crickets was the only audible sound present in the recording.

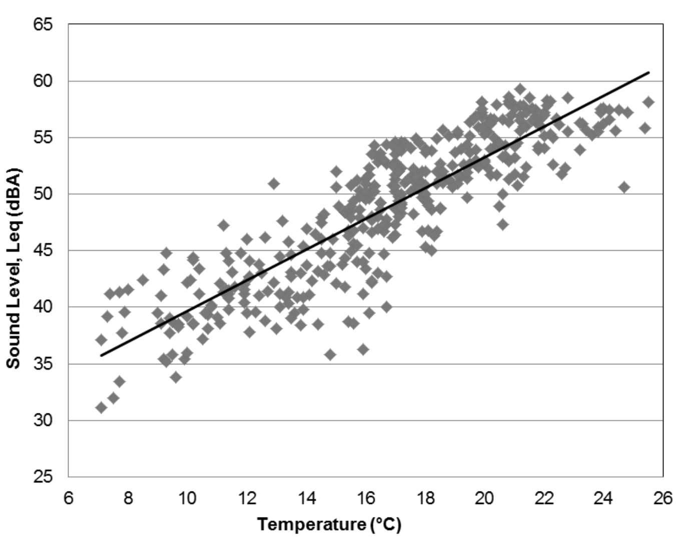

Figure 2: Correlation of Sound Level with Temperature.

3 Discussion

The acoustic environment away from urban activity is often described by the general term "sounds of nature". The many elements that comprise the sounds of nature show significant variation. This study has briefly considered only one specific component of the natural acoustic environment.

Variation correlated with temperature has been identified, although time of day and season are also factors. After sunset, the warmer temperatures correspond with higher sound level. As the temperature falls during a night there is a corresponding drop in the sound levels. The occurrence for only a part of the year, as well as the nighttime-only aspect of the sound are undoubtedly due to the life cycle and activity of the cricket to which the sound has been attributed. The exercise illustrates the complexity that would be associated with predicting the sounds of nature.

The data illustrates a general correlation between temperature and sound level. However, for each temperature there the sound levels vary by about 5 dB from the median. One of the contributing factors is likely to be the difference in temperature measurement location. The temperature measurements were taken in air, at a few meters above the ground. Crickets on the other hand are located on the ground among the grasses. A temperature measurement at ground level is likely to produce a closer correlation.

Other factors that would influence the volume of sound are the specific varieties of cricket, and the number of crickets present in an area. Additional detail from the field of entomology would be needed.

This work highlights that measurement of background sound in the presence of dominant sounds of nature requires at least a basic understanding of local influences. In this case, cricket noise far exceeded the daytime sound level. Nighttime sound levels would not satisfy the Ontario Ministry of Environment requirement for times when the background is at its lowest level [4].

4 Conclusion

A pattern of sound levels between sunset and sunrise has been attributed to the sounds of nature. They correlate with temperature, even though sound levels during the remainder of the day are not temperature correlated. Audio indicates that this is due to crickets chirping.

During the late summer measurement period, sound levels in the area do not follow a typical diurnal pattern. The quietest hourly period does not fall during the normal 0300h to 0400h period.

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FINITE DIFFERENCE TIME DOMAIN METHOD FOR ACOUSTIC WAVES IN ATTENUATE AND ABSORPTIVE MEDIUM FOR LAYERED UNDERWATER ACOUSTIC ENVIRONMENTS

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1 Présentation (Français)

Dans cet article, la méthode des différences finies (FDTD) est utilisée afin de calculer la propagation du son dans les eaux peu profondes. Le développement du SI a permet d'appliquer la dite méthode en tenant compte de l'atténuation et de l'absorption [1]. Les problèmes de l'acoustique sous marine ayant comme point de départ l'équation de Helmholtz qui a de nombreuses approximations tells que :le tracé des rayons [2],FFP (Fast field program) [3], PE (Equation parabolique) [4], ou encore la solution numérique directe des équations de Navier stockes (DNS) [5].Pour le milieu de recherche l'équation parabolique offre un meilleur compromis entre la précision et l'efficacité de ce problème [6].La validation analytique choisie est prise de knightly et de Vefring MjØlsne [7].Il s'agit d'une guide d'onde homogène de profondeur constante avec une pression de la surface de liberation et un fond rigide. Tous ces paramètres donnent un maximum de 27 modes propageant jusqu'au 84° par rapport à l'horizontale. D'autre part, afin de calculer l'amplitude de la pression (P), par rapport à la profondeur et la portée, on utilse le programme Bellhop pour tracer les pertes par transmission (TL) tout en utilisant un profil de célérité de Munk [8].Il est à noter qu'on a utilisé notamment pour les deux cas des deux couches (fluide- solide) le fameux code de Bellhop. Dans cet article, l'algorithme FDTD est validé à la fois par les solutions analytiques de la PE et des resolutions numériques fournies par le programme de Bellhop dans les fréquences supérieures à 1kHz en deux dimensions.

2 Presentation

In this paper, the Finite Difference Time Domain (FDTD) Method is proposed to calculate the propagation of sound in shallow water. The recent development of computer system enables the method to be applied in acoustics takes account of attenuation and absorption [1]. For underwater acoustics problems the starting point is the Helmholtz equation, which is have many approximations ntroduced such as: the ray trace [2], FFP (Fast field program) [3], PE (Parabolic equation) [4],or once again the direct numerical solution of Navier stockes (DNS) [5]. For the research environment the parabolic equation PE provides the best compromise between accuracy and efficiency for such problem [6]. The analytical validation is taken from knightly and from Vefring MjØlsne [7]. It involves a homogeneous waveguide of constant depth with a pressure-release surface and a rigid bottom. These parameters yield a maximum of 27 modes witch propagate up to about 84° from the horizontal.In the other hand in order to calculate the amplitude of pressure (*P*), with respect to depth and range, we use Bellhop program which is a Gaussian beam tracing to find the transmission loss (*TL*) using an Munk sound speed profile [8].We note that the bellhop popular code is especially used in the cases of two layered under water environment (fluid-solid). In this work The FDTD algorithm is validated both by analytical solutions based on the parabolic equation PE and numerical results provided by the Bellhop program at frequencies as high as 1 kHz for two dimensional problems.

3 Results

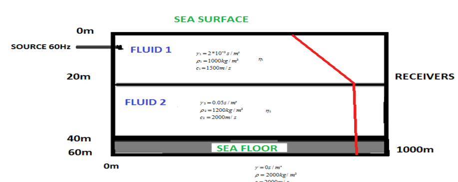

Figure 1: Geometrical of the underwater propagating

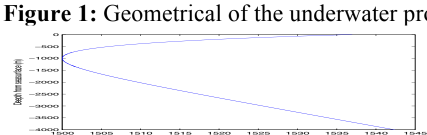

Figure 2: The Munk Sound Speed profil

3.1 Taking acount with attenuation

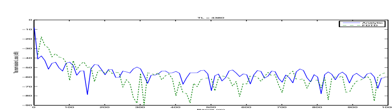

Figure 3: Acoustic FDTD for one layered medium taking account with attenuation .

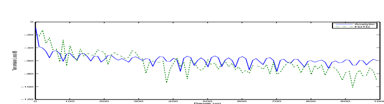

Figure 4: Acoustic FDTD for two layers (fluid-fluid) taking account with attenuation .

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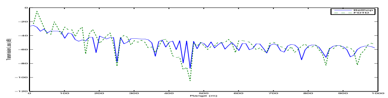

Figure 5: Acoustic FDTD for two layers (fluid-solid) taking account with attenuation .

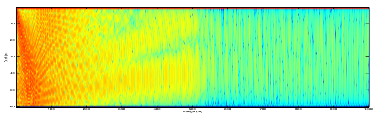

Figure 6: Acoustic field distribution for the two layers environment (fluid-solid) taking account with attenuation.

3.2 Taking acount with absorption

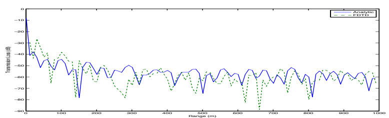

Figure 7: Acoustic FDTD for one layered medium taking account with absorption .

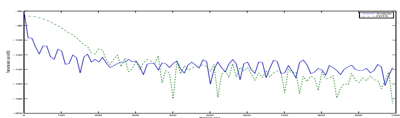

Figure 8: Acoustic FDTD for two layers (fluid-fluid) taking account with absorption.

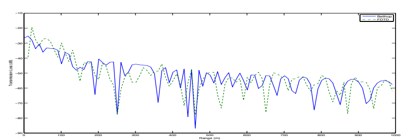

Figure 9: Acoustic FDTD for two layers (fluid-solid) taking account with absorption .

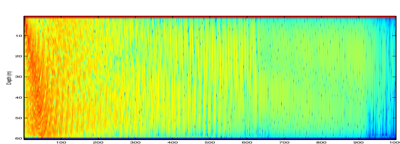

Figure 10: Acoustic field distribution for the two layers environment (fluid-solid) taking account with absorption.

4 Discussion

These results show the validity of the FDTD method. Three cases of one layered, fluid-fluid and fluid-solid two-layered medium was treated. We demonstrate that our results agree well with Bellhop method for all cases at high frequencies. Bellhop was chosen for this analysis as "it has proven to be an accurate modeling tool for high-frequency (>1 kHz) transmissions. At low frequencies our Simulation Show also a good agreement with analytical results. We have separately investigated the effect of attenuation related to water viscosity and the dissipative effect related to the absorption coefficient depending on the chemical composition of seawater.

5 Conclusion

The FDTD method is used to calculate the propagation of acoustic plane wave in shallow water with different environment. The simulation results of the transmission loss pattern are compared with analytical results of parabolic equation (PE) [8] and numerical results of Bellhop code. The FDTD is also able to visualize the propagation of the sound field in the same simulation model. We can clearly see the interference between direct wave and reflected waves.

As a future work, it is aimed to develop FDTD model with more realistic properties of the medium like roughness, temperature and pressure for calculating the TL in more realistic environments using spherical acoustic waves in three dimensions space and extend our study to more realistic sound source.

Remerciements/Acknowledgments

The authors thank Mr NIKOLAI KOLEV PHd from the Naval Academy, Varna, Bulgaria for support on this work.

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UNDERWATER ICE CRACKING NOISE MEASUREMENTS IN YELLOWKNIFE BAY, GREAT SLAVE LAKE, NEAR THE DETAH ICE ROAD

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1 INTRODUCTION

An opportunity arose on April 14, 2013 to go out on to the frozen ice road on Great Slave Lake, Northwest Territories, and attempt to measure the underwater ambient noise in Yellowknife Bay. Figure 1 shows the outline of Yellowknife Bay with the dotted line being a GPS track along the ice road. The '+' symbol denotes the location of the recording system, which was just beside the ice road. During the recording, the conditions were -12C, 6 kmh wind, the ice was 128 cm thick with a hard 20-cm snow cover, the water depth was 25 m with the hydrophone at 22 m depth, and the sun had risen 2.7 hr earlier, but was very low in the sky. The sun is estimated to be between 7 and 10 degrees above the horizon during the measurements.

A DRDC-built low-noise hydrophone and recording system were available for use along with a *hot box* to keep the electronics warm in the cold air. The recordings were made at a time when vehicular traffic on the ice road was at a minimum. Unfortunately, several vehicles did pass during the recording period. These vehicles resulted in strong overloading and the records containing the vehicle noise were not useful for the current purpose. These records were not used and have resulted in gaps in the hour-long data record.

The remainder of this document describes the equipment and presents the results of measurements of ice-cracking noise events during the one-hour recording period.

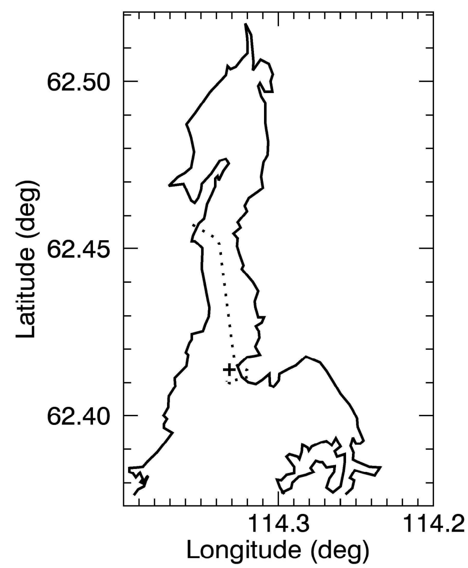

FIGURE 1 – Sketch map of Yellowknife Bay, the Detah Ice Road route (dotted line), and the measurement site location denoted by the plus sign.

2 EQUIPMENT

The sensing and recording equipment consisted of a custom built hydrophone, with a sensitivity of -203 dB, based on a Channel Industries C-5500 piezo-electric cylinder. The cylinder was closed with stiff carbon-fibre hemispherical endcaps. A 30 dB pre-amplifier with a self-noise of 1.6 nV/root-Hz was included inside the hydrophone with additional amplification located in a small pressure canister approximately 1 m distant. The amplifiers provided a 20 Hz high-pass frequency and a 30 kHz low-pass frequency filter. The minimum system gain of 60 dB was used for all recordings. The hydrophone is useful to a minimum sound pressure level (SPL) of approximately 27 dB//1μPa for frequencies from under 100 Hz to more than 20 kHz.

A National Instruments USB-9233, 24-bit analogue-to-digital (A/D) converter was used to sample the hydrophone signal at 50000 samples/sec. This A/D has a 102 dB dynamic range and a ± 5 V input range. This implies that the useable SPL range of the recording system at 60 dB gain is 55–157 dB//1μPa.

An Acer Aspire One netbook running Windows XP was programmed with NI Labview to collect a series of thirty 2-min long data files. These data files were subsequently converted to WAV format and were analyzed to provide the ice-crack rate of occurrence and spectral content.

3 RESULTS

Data files were inspected for vehicle noise contamination and interrupted recordings resulting in short files. Out of 30 files, 22 were retained for analysis. From each of these files, the number of crack events in the first 30-seconds were counted.

Cracks were manually counted using three different criteria : 1. Cracks were detected in the time series by recognizing the oscillatory nature and reverberation combined with aurally listening to ensure the signals were ice crack events, 2. A sonagram (frequency vs. time and intensity image) was created for each period and cracks were detected by searching for wideband short duration signals in the frequency band 1–5 kHz with particular attention to signals near 1.5 kHz, and 3. The number of signals strong enough to overload the system (more than 5V at the A/D) were counted in the full 2-minute file length (there were few overloads and a longer observation time was needed).

Each of these criteria result in a different rate of crack detection. The spectral method is most sensitive and results in the highest crack rate results. The time series method is strongly dependent on the general noise level and misses many cracks in the reverberation periods. The overload criterion produces the lowest crack rates and it can be assumed that the overloads are mostly counting the cracks that occur at short range from the hydrophone. The results of the three crack counting methods are shown in Fig. 2.

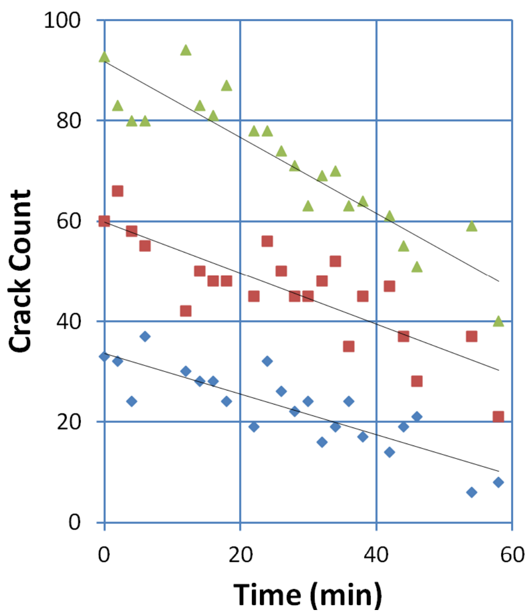

FIGURE 2 – Ice cracks as detected by three methods versus recording time. Crack events were counted using the spectral method (triangles), time-series waveform (squares), and input overloads (diamonds).

All three methods show that the crack rate is decreasing with time during the one-hour observation period. All three methods fit well with a linear decrease in the crack count versus time. The spectral method is the most sensitive and provides the best fit to the data with a linear trend. On average, the spectral and time-series methods produce consistent results, with the spectral method returning 1.5 times more crack events than the time-series method. This represents approximately 4 dB improvement for the spectral method.

Unfortunately, it isn't possible to determine the areal crack density from our single sensor as the detection range of either method cannot be determined. We also cannot determine the crack source level distribution, since the range to crack events cannot be determined. A number of the crack events do exceed the system dynamic range, so we can say that the peak source levels of the larger crack events are considerably higher than 157 dB. The spectrum levels for a period near the beginning of the hour-long data and for a second period near the end of the record are shown in Fig. 3. SPL's clearly reduce across the band as the sun rises. Most of the crack energy is below 1500 Hz, but crack energy remains appreciable up to 20 kHz. In general, noise conditions were high throughout the recording interval.

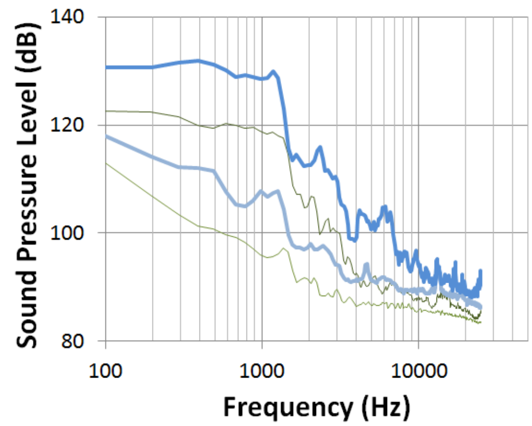

FIGURE 3 – Sound Pressure Levels for one interval near the beginning of the data record and another near the end. Thick lines (blue) represent a quiet period and a cracking noise period near the start of the data record, while thin lines (green) represent the same near the end of the record.

4 CONCLUSION

Ice-cracking events dominate the underwater ambient noise in Yellowknife Bay. The cracks were observed to occur at a rate of 1–2 a second during our 1-hour observation. The crack rate constantly reduced during the observation and it is presumed that this is due to the ice adjusting to the impact of solar heating. Detection of ice-crack sounds is not a simple matter, but three straight-forward approaches were described. The spectral content method was the most sensitive.

Unfortunately, location and areal density of cracks cannot be determined from our single omnidirectional hydrophone data. We can say that a range of ice-crack source levels are to be expected and a peak instantaneous level well above 157 dB//1μPa has been repeatedly observed.

ACKNOWLEDGEMENTS

Thanks to Joint Task Force North and 1 Canadian Ranger Patrol for supporting this work.

Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Model-Data Comparisons For Transmission Loss And Reverberation During Trex13

Sean Pecknold, Dale D. Ellis, Cristina D.S. Tollefsen

TREX13 (Target and Reverberation Experiment 2013) was a sea trial that took place in a shallow-water (approximately 20 m deep) environment near Panama City, Florida, in the Northeastern Gulf of Mexico during April and May of 2013. Sets of acoustic transmission loss and reverberation data were acquired on a fixed horizontal array for frequencies in the range of 1800 Hz to 3600 Hz. Several of these data sets involved transmission along a "reverberation track", where the fine-scale bottom roughness was well characterized by a multi-beam bathymetric survey. The effects of boundary roughness and scattering from some of the features along this track are investigated via a comparison of the acoustic data to parabolic equation and normal-mode propagation modeling of the transmission loss and reverberation.

Perceptual Feature Extraction For Target Classification On A Biosonar Dataset

Tara Joy LeBlanc, John Fawcett

In the field of underwater unexploded ordnance (UXO) detection and identification, the current state of the art relies heavily on imaging sonars with frequencies on the order of 100's of kHz. These sonars generate detailed images of the seafloor, which are then analysed via shape-based target recognition algorithms for the presence of targets. In areas where clutter objects possess similar dimensions to targets of interest, this approach becomes overwhelmed with false alarms. Low-frequency (10's of kHz) broadband sonar is emerging as the technology to mitigate this problem, because these lower frequency acoustic signals can generate returns from both elastic effects and a target's internal structure. Low-frequency broadband sonars aren't imaging sonars in the conventional sense and the challenge is to identify targets using non-shape-based methods. The Biosonar, developed by Hydrason Solutions of Heriot-Watt University, is a sidescan sonar with a nominal frequency range of 30 – 130 kHz and dual centre frequencies of either 90 kHz or 60 kHz. Trials were conducted using this Biosonar to survey a series of cylindrical targets with identical structure yet differing internal fillers and the current work analyses this data. Previously, the authors calculated signal features based on the human perception of timbre for the subset of data corresponding to the lower centre frequency (60 kHz) and applied the Fisher discrimination method with good results. The current work extends that approach to the data corresponding to the higher centre frequency. The result is a marked difference in the Fisher distribution between targets of the same fill but differing centre frequencies. These differences in frequency content between the two datasets are discussed along with how they impact the perceptual features. Further, a possible scheme to fuse the features between the different centre frequency datasets is explored.

[SPECIAL SESSION] HEALTH CARE FACILITIES

FLOOR VIBRATION CONSIDERATIONS FOR SENSITIVE EQUIPMENT IN HOSPITAL, MEDICAL, PHARMA AND LABORATORY FACILITIES

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1 Introduction

Advances in materials, construction, and design technology have led to the construction of lighter floor systems, which are more susceptible to footfall-induced vibration. Meanwhile, improvements to imaging technology, including magnetic resonance imaging (MRIs) and electron microscopes, have led to devices which have increasingly stringent vibration requirements to ensure optimal performance. As a result, it is becoming more common for floor vibrations to be the governing factor in the design of the structural system.

The primary source of vibration in most facilities is human activity. As people walk, the impact from each footfall induces floor motions that may easily transmit to nearby spaces. Quantifying vibration from walking, whether through measurement of existing spaces or numerical predictions for guiding the design of a new facility, is a complex task. This task is complicated in part by the availability of a number of vibration measurement and prediction methodologies, each associated with both similar and unique assumptions.

In this paper we discuss generic and specific vibration criteria that are commonly used in international practice for sensitive spaces. Several predictive models for footfall-induced vibrations are discussed that apply to both steel and concrete construction.

2 Vibration Criteria

For human comfort, vibration criterion is normally expressed as the root mean square (RMS) response of each one-third octave band from 1 Hz to 80 Hz [1]. For sensitive equipment, the criteria may be expressed in one-third octave bands, or other formats, including power spectral densities, peak-to-peak levels, etc. Over the past 25 years, generic vibration limits have been developed which provide frequency-dependent sensitivities for wide classes of equipment, and are used extensively in design of healthcare and research facilities [2]. These vibration criterion (VC) curves are internationally accepted as a basis for designing and evaluating the performance of vibration sensitive equipment and the structures that support them. The VC curves range between Workshop (least stringent) through VC-G (most stringent). See Table 1 for a list of descriptions of use for spaces meeting these criteria levels.

These curves were originally based on the ISO 2631-2 (1989) [3] base curve for human response to whole body vibration, which is the threshold of human perception, but have since evolved. The ISO base curve is often referred to as the ISO-Operating Theatre criteria. The VC curves should not be used to replace manufacturers' specifications for vibration requirements, but are beneficial where manufacturers' specifications are non-existent, incomplete, or where specific equipment has not yet been selected.

Vibration Criteria	Description of Use
Workshop (800 μm/s)	Distinctly perceptible vibration. Appropriate to workshops and non-sensitive areas.
Office (400 μm/s)	Perceptible vibration. Appropriate to offices and non-sensitive areas.
Residential Day (ISO) (200 μm/s)	Barely perceptible vibration. Maximum recommended for general sleep areas. Usually adequate for computer equipment.
Residential Night (140 μm/s)	Appropriate for most sleep areas such as hospital recovery rooms.
Operating Theatre (100 μm/s)	Threshold of perceptible vibration. Suitable in most instances for surgical suites, catheterization procedures and microscopes to 100 X magnifications.
VC- A (50 μm/s)	Adequate in most instances for optical microscopes to 400X, micro-balances, and optical balances.
VC- B (25 μm/s)	Micro-surgery, eye surgery and neurosurgery, CT, CAT, PET, fMRI, SPECT, DOT, EROS.
VC- C (12.5 μm/s)	Appropriate for MRIs, NMRs, standard optical microscopes to 1000X magnification, and moderately sensitive electron microscopes to 1μm detail size.
VC- D (6.25 μm/s)	Suitable in most instances for demanding equipment, including may electron microscopes (SEMs and TEMs) at more than 30,000X magnification and up to 0.3 micron geometries, and E-beam systems.
VC- E (3.12 μm/s)	Assumed to be adequate for the most demanding of sensitive systems including long path, laser-based, small target systems, and systems working at nanometer scales.

Table 1: Vibration criteria descriptions of use.

The Guidelines for Design and Construction of Hospitals and Outpatient Facilities (FGI 2014) [4] provides limits for floor vibration in several types of areas in healthcare facilities caused by footfalls, which are consistent with those found in Table 1. The CSA Standard Z8000-11 (Canadian Health Care Facilities) [5] also refers to these vibration criteria levels. These two documents require the use of the American/Canadian Institute of Steel Construction (AISC/CISC) Design Guide 11 (DG11) [6] to provide footfall-induced vibration predictions.

3 Predicting Footfalls

3.1 American/Canadian Institute of Steel Construction Design Guide 11

The modeling technique outlined by the American Institute of Steel Construction (AISC) Design Guide 11 has been extensively used in North America for the past 15 years. For human comfort, the AISC method assumes the floor behavior is governed by the resonant response of the fundamental floor mode. A steady-state floor response is calculated assuming the walker and vibration-sensitive receptor are both located at the position of the maximum modal displacement (center of the bay) to produce a worst-case response. The natural frequency of the floor is estimated from the maximum static deflection of the bay under consideration due to the acting dead and live loads. Such a simplistic method cannot easily predict the natural frequency when the floor layout is irregular, such as would occur around shafts and/or in non-rectangular buildings.

For areas with sensitive equipment, the AISC considers three walking speeds: 50, 75, and 100 steps per minute. An empirical force coefficient is estimated as a function of the walking pace, and the weight of the walker. Simple beam theory is employed to calculate the floor deflection, and the natural frequency. The floor response velocity is then calculated as a function of the force coefficient, floor natural frequency, and floor deflection.

3.2 Steel Construction Institute and Concrete Centre

The methodologies proposed by The Steel Construction Institute (SCI P354) [7] and The Concrete Centre (CCIP-016) [8] for calculating footfall-induced floor vibrations are similarly derived. The methods recommend utilizing a Finite Element (FE) model to predict the mode shapes and natural frequencies of the floor. FE modeling is advantageous since it allows the calculation of many mode shapes, any of which can contribute significantly to the vibration of a specified floor region. Moreover, irregular floor features (including non-rectangular bays, shafts, and different beams or slab thicknesses from bay-to-bay) can easily be accommodated by the FE model.

The response of each mode is assumed to be either resonant or impulsive, depending on the associated floor frequency. Empirically determined frequency-dependent dynamic load factors are used to excite the floor along probable walking paths as determined from the architectural drawings. The loading conditions are then used to develop a time series response for each mode. The time series responses for each mode are then linearly superimposed to produce a total floor response at any receptor point on the floor. A range of realistic walking frequencies and walking paths are considered to determine the governing floor response. Spectral analysis can be performed on the predicted time-series responses to express the floor behavior in a format appropriate for comparison with the relevant criteria.

Refer to [9] for a comparative study on the accuracy of these methods.

4 Conclusion

This paper presented an overview of human-induced vibration in concrete and steel buildings. The impact of these vibrations can have a detrimental effect on the performance of sensitive equipment and impact the occupants through annoying and potentially alarming motions. There are several established and evolving criteria for determining acceptable levels of vibration which range from far below perceptibility to motions that are very noticeable.

Several methods for predicting the levels of human-induced vibrations are in widespread use internationally, with three of the more common methods being the AISC Design Guide 11, SCI P354 and CCIP-016. The AISC method is based on empirical factors, and is most useful for quickly predicting motions on floors that have simple and repeated layouts across all bays, while the SCI and CCIP methods depend on the development of a Finite Element Model in order to capture the more complicated behaviour of complex structures.

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FGI GUIDELINES FOR HEALTHCARE ACOUSTIC DESIGN

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1 Introduction

The Facility Guidelines Institute is responsible for the production of their industry-standard document focused on the design and construction of health care facilities in the United States. This once-singular document has been divided into two separate volumes since the latest (2014) revision cycle : *Guidelines for the Design and Construction of Hospitals and Outpatient Facilities*, and *Guidelines for the Design and Construction of Residential Health, Care, and Support Facilities* (hereafter referred to as FGI 2014). Currently, 42 U.S. states use at least one edition of the *Guidelines* in some form into their building codes and they have been read in 60 countries [1].

The *Guidelines* have been available in the United States in multiple iterations beginning in 1947 as the *General Standards* [2]. Today, the *Guidelines* are edited and re-issued every four years by a a multi-disciplinary group of experts related to healthcare facilities. Due to such issues as patient privacy and evidence linking hospital acoustics and patient recovery, acoustical criteria for healthcare facilities was formally introduced into the 2010 cycle. They are included in both volumes of the *Guidelines* and cover six categories, which will be explained in detail in the following sections of this paper :

1. Site exterior noise
2. Acoustic surfaces
3. Room noise levels
4. Interior wall and floor/ceiling constructions
5. Speech privacy
6. Building vibration

2 FGI Guidelines Acoustical Criteria [3, 4]

2.1 Site Exterior Noise

FGI 2014 requires that the sound isolation provided by the exterior shell (including the exterior wall/window assemblies, penetrations, etc.) of health care facilities be designed to result in appropriate interior noise levels. Exterior noise transmitted into the building and noise produced by the facility reaching nearby receptors (neighbors) are considered.

This section classifies exterior site noise into one of four categories based on the site exposure and provides prescriptive building envelope Outdoor-Indoor Transmission Class (OITC) ratings. The noise exposure categories are determined by the day-night average sound level (L_{dn}, dB) and the average hourly maximum sound level (L_{01}, dBA). A separate, non-prescriptive, resource is also given to approximate the site noise exposure by the distance to nearby transportation noise sources.

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2.2 Acoustic Surfaces

In order to foster a pleasing acoustical environment for patients and staff, the FGI 2014 sets minimum average room absorption coefficients for multiple types of healthcare facility spaces based on their use. Any material used must also satisfy all infection control/cleaning requirements as defined by the facility.

Rather than requiring specific finishes to be used, FGI 2014 defines acoustical performance based upon the average sound absorption coefficient, which is calculated as the sum of all boundary areas in the room first multiplied by the sound absorbing performance of each material in the room and divided by the sum of the boundary areas.

2.3 Room Noise Levels

Noise from building mechanical systems is an important aspect of the acoustical environment and contributes to the overall comfort of both patients and facility staff. Since several metrics exist to define noise levels in a room, FGI 2014 presents the maximum criteria for noise in interior occupied spaces based on functionality in terms of NC, RC (Neutral), RNC, and dBA. FGI 2014 also clarifies that the criteria outlined refer only to maximum building mechanical system noise rather than overall interior noise levels due to occupant and/or medical equipment noise. The presentation of maximum noise criteria, rather than the minimum-maximum criteria presented in FGI 2010 is new to FGI 2014.

2.4 Interior wall and floor/ceiling constructions

Interior wall and floor/ceiling constructions must provide adequate sound isolation for patient and staff comfort as well as to meet patient privacy requirements. FGI 2014 sets forth minimum required composite sound transmission class (STC_C) performance based upon the function of adjacent spaces.

Since the 2010 cycle, the FGI 2014 requires the STC_C for patient, consultation and exam rooms adjacent to a corridor (including the door) to be 35, excluding the door, rather than being STC_C 35 with a closed door. This implies that the performance objective is now assigned to the partition itself. FGI 2014 also states that doors are not required to be sound sealed and the use of higher performing doors, full-perimeter gasketing, and bottom seals will be left to the discretion of the facility. These changes were made in response to concerns that the composite rating including doors required door hardware that may not be compatible with infection control and cleaning requirements.

2.5 Speech Privacy

United States law through the Health Insurance Portability and Acountability Act (HIPAA) requires facilities to safeguard patients' private information [5]. FGI 2014 requires that spaces be designed to meet speech privacy goals using one of four speech privacy metrics : PI, AI, SPC and SII. By not specifying exact design criteria to meet speech privacy needs, greater flexibility in the design is allowed based upon the needs and expectations of the specific facility.

2.6 Building Vibration

Mechanical, Electrical and Plumbing Equipment Vibration

Most building mechanical equipment generates vibrations that can be transmitted to the building structure. To avoid structure-borne transmitted sound, and to minimize impact on human comfort or sensitive equipment, FGI 2014 states that all rotating and vibrating components of the building systems should be isolated as recommended in the most current Applications Handbook of the American Society of Heating, Refrigerating, and Air-Conditioning Engineers (ASHRAE).

Structural Vibration

FGI 2014 requires that the building structure must be designed so as to avoid footfall vibration levels in excess of specific values. In addition, because hospitals and healthcare facilities are likely to have sensitive equipment (i.e. MRI, microscope), the building must be designed so that structural vibrations do not exceed the limits recommended by the equipment supplier.

Structure-borne Sound

FGI 2014 requires that structurally transmitted sound may not exceed the airborne room noise levels discussed in the section, "Room Noise Levels" and also requires vibration isolators to be used on sources of structurally borne sound when necessary.

3 Continual Improvement Process

In order to stay current with the latest research and changes in healthcare laws, the *Guidelines* are edited and re-issued every four years following an open, formal, Continual Improvement Process. This work is done by the FGI's Health Guidelines Revision Committee (HGRC), which is composed of a multidisciplinary group of experts related to healthcare facilities.

The acoustical content is the responsibility of the FGI Acoustics Working Group, of which the author holds membership (as co-chair of education). This group authors and edits the *Sound & Vibration Design Guidelines for Health Care Facilities* which serves as the sole acoustical reference for the *Guidelines*.

The revision cycle begins shortly after publication of the current edition of the *Guidelines* with a period for public submission of proposed changes. Then, the HGRC considers the public proposals and proposals prepared by HGRC members and subcommittees, and votes on whether to accept, accept with modification reject, or reject them.

The result of this committee work is a draft manuscript for the next edition of the *Guidelines* which is posted on the Internet for public review and comment. After this review period the HGRC meets again to consider the comments on the draft document, and finalizes the content for the next edition. After this meeting, a final draft is reviewed by the Steering Committee and then submitted to the HGRC for final approval by ballot. Subsequently, the next edition of the *Guidelines* is published.

4 Conclusions

Both the *Guidelines for the Design and Construction of Hospitals and Outpatient Facilities*, and *Guidelines for the Design and Construction of Residential Health, Care, and Support Facilities* serve as an industry standard documents for the design and construction of health care facilities in the United States. Their acoustical criteria, brought on by both patient privacy laws and evidence of detrimental effects of hospital noise on patients and staff, help enforce the importance of proper acoustical design in healthcare facilities. As evidence of the benefit of proper acoustical treatment in healthcare facilities grows, it is expected that incorporation of the latest acoustical issues will be incorporated into future editions of the *Guidelines*.

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Hospital noise methodology and survey results

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1 Introduction

Hospital noise is an ongoing issue. With high amounts of activity and equipment, many hospitals are producing noise levels that can impede patient healing and affect staff [1, 2]. Therefore, it is to little surprise that noise is a commonly low-performing category in patient satisfaction surveys.

In the United States, with the enactment of the Patient Protection and Affordable Care Act in 2010, Federal reimbursements to hospitals are dependent, in part, upon the performance of the Hospital Consumer Assessment of Healthcare Providers and Systems (HCAHPS) patient satisfaction survey. This survey accounts for patients' perception of the noise level in terms of the area around their room being, "quiet at night" which has historically shown poor performance [3].

Because of the widespread use and importance of the HCAHPS system in the United States, it was of interest to compare patient satisfaction scores with their associated noise levels. Acoustics By Design (ABD) has performed several comprehensive noise assessments of healthcare facilities and gathered the associated patient satisfaction data. The purpose of this session is to explain the HCAHPS survey, describe the hosptial noise survey methodology developed by ABD, and compare selected results of ABD's noise surveys with the facilities' patient satisfaction scores.

2 HCAHPS

The Hospital Consumer Assessment of Healthcare Providers and Systems (HCAHPS) is a powerful comparative tool within the United States because it serves as a nationally standardized and publically available hospital rating system. It was developed in part by the Centers for Medicare & Medicaid Services (CMS) and the Agency for Healthcare Research and Quality (AHRQ). It was first implemented in 2006 with the first public result reporting period in 2008 [4].

Relevant to hospital noise the HCAHPS survey asks, "During this hospital stay, how often was the area around your room quiet at night ?" which is answered as "never", "usually", "sometimes" and "always." The data is presented to the public as the percentage of patients, "who reported that the area around their room was, 'Always' quiet at night". Recent findings show that less than 60% of respondents patients answer "always", which reflects significantly poorer performance than the other categories [3].

The enactment of the Patient Protection and Affordable Care Act in the United States changed Federal reimbursements of qualified hospitals to be based on a Value-Based Purchasing (VBP) plan which evaluates the performance of HCAHPS scores starting in October 2012. This action has been shown to be a significant influence in the adoption of noise mitigation in hospitals. In a recent comprehensive assessment of patient experience, it was found that 65% of respondents indicated "pay for performance" issues as the main reason to reduce noise [3].

3 Hospital Noise Survey Methodology

Acoustics By Design (ABD) has developed a comprehensive hospital noise survey methodology that has been conducted in multiple hospitals across the Midwest region of the United States.

3.1 Measurements

ABD has integrated multiple techniques in performing hospital noise surveys that consist of both a long term measurement session (to capture the broadband sound level and describe the overall noise over time and in different locations) and a short term measurement session (to characterize disturbing singular noise source events).

Measurement locations are chosen in an interdepartmental planning session with hospital staff to ensure measurements are collected in representative locations. This helps to gather information on which areas are perceived as to be quieter or louder and other operational factors such as cleaning/food service schedules, where staff tend to congregate, and the patient room door policy. Typically, measurement locations are at several representative locations including corridors, nurse's stations, private patient rooms, semi-private patient rooms and unoccupied patient rooms.

Long Term Noise Monitoring

In congruence with the methodology of similar current research [5–7], ABD has chosen to perform long-term broadband measurements with Larson Davis 703 and 706RC dosimeters. The meters are programmed for slow response and 1-second sampling of LA_{eq}, L_{max}, L_{min}, and L_n. 1-second data is used to evaluate how much the sound level varies and how quickly. It is also used for correlation with the short term noise source data (described below).

Long-term measurements are conducted over a 48-hour period and typically cover one weekend day and one weekday to attempt to account for variations in census counts. In an effort to reduce staff behavioral changes, installation typically occurs several days prior to the actual measurement session and the staff is not informed on when the meters begin measuring.

Short Term Noise Monitoring

In addition to the long-term broadband noise monitoring described above, short term measurements are collected during

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the long term measurement period in order to characterize the level and frequency content of many identified disturbing singular noise source events.

These measurements collect short-term (typically about 30 seconds to 1 minute in duration) octave band sound level data acquired at a normalized distance of 1 meter from each source using a Larson Davis 824 sound level meter. The selected noise sources are based on observations during hospital tours as well as staff surveys. Common sources include patient bed alarms, cart noise, the prescription delivery system, the paging system, and EVS equipment.

4 Results

The following describes the noise survey results and patient satisfaction scores of one facility in the Midwestern United States conducted before and after implementation of ABD's noise mitigation recommendations. Table 1 below shows the L_{eq} and patient satisfaction scores of the different departments of the facility before and after mitigation. The satisfaction scores are reported as the percentage of patients who perceived the noise in and around their room to be "favorable" which is assumed to be similar to the HCAHPS survey question metric.

Unit	L_{eq} before (dBA)	Satisfaction before (% favorable)	L_{eq} after (dBA)	Satisfaction after (% favorable)
A	57	66	53	81
B	59	70	53	76
C	56	69	52	73
D	57	68	53	76

Table 1: L_{eq} and patient satisfaction scores (percent of patients that described the noise in and around their room as "favorable") before and after implementation of noise mitigation. [8]

According to Table 1, before mitigation, the average noise levels within the different departments ranged from 56-59 dBA over a 48-hour period. For comparison, a common hospital design reference, the FGI Guidelines (2014 cycle), recommends a maximum background noise level of 45 dBA within patient rooms (produced by the building mechanical system). However, because it was not feasible to reduce noise levels to those levels, recommendations given to the client were designed to minimize the overall background noise level along with reducing the frequency and level of disturbing impulsive noises, with the end goal of achieving better "acoustic comfort".

After mitigation, the decrease in overall L_{eq} ranged from 3-6 dBA. While the level of the "after" measurements are still higher than what is recommended by the FGI guidelines, the patient satisfaction scores in the departments of interest increased from 4-15%. In addition, anecdotally, the hospital reported that, overall, patients changed their perception of the treated spaces from being "noisy" to being "quiet". These data suggest a preliminary relationship between the decrease of hospital noise level with the increase of patient satisfaction scores [8].

5 Conclusions

The Hospital Consumer Assessment of Healthcare Providers and Systems (HCAHPS) patient satisfaction survey provides a standardized reporting method across the United States whose results include a category for the perception of hospital noise. Because federal reimbursements to many hospitals across the United States are now dependent on the performance of the HCAHPS survey and because hospital noise is one of the poorest performing categories, it is expected that there will be an increase of interest in hospital acoustics studies.

Acoustics By Design (ABD) has performed multiple hospital noise studies across the United States and have developed a comprehensive and cost effective methodology. One particular study study involved performing identical measurements before and after the implementation of noise mitigation techniques. The results from this study show that after mitigation the background noise level was reduced by 3-6 dBA, while patient satisfaction scores of the "noise in and around the room" increased from 4-15% favorable. This preliminary evidence seems to indicate a relationship between increasing patient satisfaction by reducing noise levels.

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Abstracts for Presentations without Proceedings Paper
Résumés des communications sans article

Acoustic Design Considerations In Modern Hospital Design

John C Swallow, Michael J Wesolowsky

The new FGI Guidelines and Consultant's experience in Hospital Acoustic design are reviewed.

Evaluation Of Objective And Subjective Acoustical Quality In Office-Type Health-Care Facilities

Leila Scannell, Murray Hodgson, Denny Ng, Juliette Rauscher

Health-care facilities include many non-clinical office-type spaces for workers; the role of acoustics in occupants' perceptions of these spaces has been underexplored. The current study assessed the reverberation time, speech intelligibility, speech privacy, noise levels (i.e., in occupied as well as unoccupied spaces), noise isolation, and DL2 in over 20 offices belonging to two local health authorities. The acoustical conditions were compared to design criteria to evaluate their acceptability. Physical acoustical results were examined according to various design features such as office type (e.g., open plan vs private office), closed office partition design, and cubicle partition height. In addition, a subjective survey administered via an online questionnaire tool assessed office workers' perceptions of their environments in general, and satisfaction with their acoustics. Self-reported productivity, well-being, and health outcomes were also captured. A multiple regression will be used to identify which physical acoustical features are the strongest predictors of the subjective outcomes. Taken together, this knowledge will inform the decision-making of designers and facilities management for upgrades and future design projects. Because relatively few studies have examined these objective-subjective interrelations in health-care facilities, the project will also contribute to the scholarly literature on acoustics and workplace satisfaction.

[SPECIAL SESSION] HEARING ACCESSIBILITY AND AGING

BEYOND HEARING AIDS: TECHNOLOGIES TO IMPROVE HEARING ACCESSIBILITY FOR OLDER ADULTS WITH HEARING LOSS

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1 Presentation

Hearing impaired older adults can experience significant difficulties accessing auditory information using their hearing aids alone. Listening in noisy environments, in poor acoustics and from a distance can present challenges even with advanced adaptive directional microphones and the signal processing strategies found in today's hearing aids. In addition, difficulties hearing on their telephone, cellphone, listening to TV broadcasts, computers, etc., can increase the frustration levels of many older adults, impacting quality of life and overall hearing health.

Each individual presents with a unique set of hearing needs and challenges. Factors such as the degree and configuration of their hearing loss, speech understanding and central processing capabilities, cognitive state and other factors contribute to the individual's communicative condition. Unless the clinician fully understands the auditory abilities and communication needs of the older adult, the recommendations for hearing assistive technologies could be inappropriate or at best only partially resolve the challenges facing the individual.

It is beyond the scope of this presentation to discuss in detail the various components that make up an auditory needs assessment; however, some key measures would include assessing hearing thresholds, speech understanding in noise (SIN), cognitive screening, a measurement of manual dexterity, and a detailed understanding of the communication needs and hearing goals of the individual.

Only after a complete picture of the individual's needs and challenges has been determined can the clinician then present appropriate technology options from an ever increasing array of devices. As noted by Sergei Kochkin (2007), a hearing aid user's satisfaction is directly related to the number of listening environments they can successfully participate in. This so called "multiple environmental listening utility" or MELU is a key focus for the hearing aid industry in designing products that will be well received by the market.

Rather than focusing on traditional technologies, such as induction loops, FM and infrared systems which have been available for many years, this presentation will look at newer technologies and devices offered by hearing aid manufacturers to enhance the overall performance of their hearing aids. These devices are relatively new to the market having been released in the past few years. They offer improved access to speech in challenging situations as well as better access to telephones, televisions, audio from computers and other like devices.

2 Recent developments in hearing aid accessibility technologies

Major manufacturers of hearing aids have in the past five to six years developed technologies to enhance the connections of their hearing aids to cellphones, televisions, music players, computers, and Bluetooth transmitting devices. These devices stream audio signals to the hearing aids using transmission signals like Bluetooth, 900 MHz or 2.4 GHz to connect the user's hearing aids wirelessly. GN Resound, Oticon, Phonak, Siemens, Starkey, Unitron, Widex and others may take different approaches, but all share the same goal, to provide better connectivity for the individual who is hard of hearing to modern communication devices and to gain better access to sources of entertainment.

2.1 Streamers

A number of manufactures have developed streaming devices to enhance the connection of the individual's hearing aids to cellphones. Based on a Bluetooth connection these streamers use various approaches to bring the caller's voice to the hearing aids wirelessly. This is done either through an induction loop or by streaming directly to the circuitry of the hearing instrument.

Phonak ComPilot Streamer with RemoteMic

The streamers all provide wireless "hands free" connectivity to cellphones using the protocols employed in Bluetooth ear pieces worn by the general public. The streamers provide binaural access to the caller's voice. This binaural access has the potential to greatly improve the user's speech understanding. In a study of hearing aid users in seven different telephone listening conditions Picou and Ricketts (2011) identified that when subjects had binaural access to a phone message the highest levels of understanding were observed, improvements of up to 35% over the monaural condition were noted.

Manufactures GN Resound and Starkey, working with Apple Inc., have developed apps that can directly connect iPhones to their hearing aids eliminating the need for an intermediary streaming device.

Streamers also have the ability to stream music and audio using A2DP Bluetooth protocols. They also act as a remote control for the individual's hearing aids. For individuals with dexterity and/or vision issues, or who purchase hearing aids without onboard controls, this can provide easier access to volume adjustments and program changes thus increasing the opportunity to make use of special listening programs available in their hearing aids.

2.2 Remote microphones

All of the manufacturers mentioned above have developed small remote microphones that either work with a streamer or have the ability to connect directly to the hearing aids. These remote microphones can be used to reduce the impact of distance and low levels of background noise. A good application for the remote microphone is in the car or in a social situation at a dining table. These low cost microphones have limited signal processing, with the majority using omni directional microphones. Due to limited signal to noise ratio (SNR) improvement for high levels of noise or transmission over extended distances the use of a personal FM or Roger system would be preferable.

2.3 TV and computer/tablet access

These same manufacturers have developed a number of devices to provide enhanced access to audio signals from TVs, computers, music players, tablets, etc. as mentioned previously these devices can stream directly or in conjunction with a streamer, the audio signal from the source to the hearing aids. For example Oticon, Phonak and Siemens all have TV devices that work with their streamers to deliver the TV audio to the users hearing aids.

Phonak TVLink with ComPilot

Manufactures GN Resound, Starkey and Widex have TV devices that can stream the TV audio directly to their compatible hearing aids. Most systems stream in stereo which can further enhance listening enjoyment. These devices can also be used to connect to other audio sources such as a computer, music player or tablet and transmit the audio signal of interest.

2.4 Landline phones

Phonak and Widex have launched cordless phones that will automatically connect their compatible hearing aids binaurally to the caller’s voice as soon as the phone is held in close proximity to the head. As mentioned previously the binaural connection can greatly improve speech understanding. This type of phone also allows for greater freedom of phone placement. The older adult does not have to achieve or maintain a particular position of the phone relative to the hearing aids in order to maintain a connection.

2.5 Roger

A newer technology has been launched by Phonak called Roger (aviation term meaning “message received and understood”) which features adaptive, wireless remote microphone transmission on the 2.4 GHz band. Roger audio signals are digitized and packaged into very short digital bursts of code that are broadcast several times using different channels between 2.4000 and 2.4835 GHz. Frequency-hopping between channels, in combination with these repeated broadcasts creates a robust transmission and avoids interference issues that can occur with personal FM system.

Phonak Roger Pen transmitter w/ charging/audio input stand

In a recent study of hearing aid users Linda Thibodeau (2014) compared Roger to traditional FM and adaptive FM systems. She found improvements in speech understanding of 54% and 35% respectively in high levels of noise (figure 1).

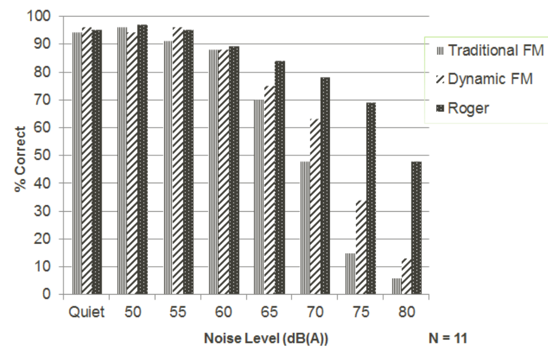

Figure 1: HINT percent correct scores for total words correct as a function of noise level for Traditional FM, Dynamic FM and Roger (adapted from Thibodeau).

3 Conclusion

Newer hearing aid assistive technologies available from hearing aid manufacturers illustrate the growing enhancement of hearing accessibility now being afforded older adults. Not all devices preform at the same level or are appropriate for all individuals. Careful assessment of auditory needs along with adequate training and support are critical to the successful implementation of these technologies.

Acknowledgments

Photos courtesy of Phonak

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AGING VOICES AND SPEECH INTELLIGIBILITY: IMPLICATIONS FOR COMMUNICATION BY OLDER TALKERS

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1 Introduction

The voice changes with normal aging. For instance, older adults may have more extreme values on measures of irregularities in fundamental frequency (jitter) and intensity (shimmer) than the voices of younger adults [1]. Listeners perceive such voices to be rougher [2] and perform more poorly on word recognition when speech has been synthetically jittered [3]. Given that some older adults have voices that naturally contain more of these irregularities, it is possible that their voices would be more difficult to understand in challenging noisy situations, and when their communication partners are experiencing age-related changes in auditory processing. To investigate the effects of voice quality on intelligibility, we recorded the speech of three older talkers who had different amounts of vocal jitter and shimmer. We predicted that intelligibility would relate to the voice quality of the talkers.

2 Experiment 1

2.1 Method

2.1.1 Talkers

Three older adult females between 68 and 74 years of age were selected from a database of recordings of healthy adults [1]. They were native Canadian English speakers who reported that they were in good health. They had pure-tone audiometric thresholds ≤ 25 dB HL from 250 to 3000 Hz, and inter-aural differences in thresholds ≤ 15 dB from 250 to 8000 Hz. These talkers were selected based on the percentile ranks of their values of jitter (local) and shimmer (local) within their age and gender group (see Table 1).

	Jitter (%)		Shimmer (%)	
Talker	Mean	Rank	Mean	Rank
Worst	0.85	16	4.87	12
Mid	0.29	65	3.36	25
Best	0.17	90	1.01	90

Table 1: Values of three voice acoustic measures and percentile ranks for the three selected talkers. Worst = poorest voice quality; Mid = medium voice quality; Best = best voice quality.

2.1.2 Stimuli recording

Talkers recorded the Northwestern University No. 6 (NU6) word recognition test items [4] in a single-walled IAC booth using a Sennheiser Linear E825S microphone placed 5 cm away from the lips. Tucker-Davis Technologies System III hardware was used. Items were monosyllabic target words following the carrier phrase "*Say the word* ____". For the recording sessions, sentences were presented on a monitor at a rate controlled by the experimenter. Talkers were instructed to "read the sentence aloud in your normal, most comfortable voice" and they took frequent breaks. Prior to each recording session, a sample of recordings from the Words in Noise (WIN) test [5] spoken by a professional talker was played to demonstrate an appropriate speaking rate; five practice sentences were spoken by each talker. The four NU6 lists were recorded over two days in different orders on each day to yield four tokens of each of the 200 sentences. During editing, the RMS energy of each sentence was equated to 0.05 Pa using a custom MATLAB program.

2.1.3 Listeners

Listeners were 16 younger adults (M=18.4 years, SD=0.6) who had learned English before the age of 5 years in an English-speaking country and had pure-tone thresholds ≤ 20 dB HL from 250 to 8000 Hz, with no significant inter-aural difference in thresholds. Listeners gave informed consent and received course credit for participating.

2.1.4 Procedure

Listeners were tested on four NU6 lists, each spoken by a different talker (3 females + the professional WIN talker), while seated in a double-walled IAC booth. Sentences were presented monaurally over TDH-50P earphones at 70 dB SPL and mixed with multi-talker babble from the WIN test at +1 dB SNR. Participants were instructed to report the last word of each sentence and guessing was encouraged with no time limit on responding. The combination and order of NU6 lists and talkers was counterbalanced across listeners. Before data analysis, listener scores were transformed from raw scores to rationalized arcsine units (RAU) [5], and all post-hoc t-tests were conducted with Holm-Bonferroni correction. Acoustic measures of stimuli were obtained using the Praat speech analysis program [6].

2.2 Results

Surprisingly, the talker with the poorest voice quality was as intelligible as the talker with the best voice, and both were more intelligible than the talker with the medium-quality voice, while the professional talker was the least intelligible of all (Figure 1). This pattern of results was confirmed by a within-subjects ANOVA, showing a significant effect of talker, $F(3, 45) = 130.5$, $p < .001$. The mean correct word recognition scores for the talkers with the best and worst voices did not differ significantly ($p = .31$); their scores were different than that of the talker with the medium-quality voice (p's < .001), and the score for the professional talker was different from all other talkers (p's < .001).

2.3 Acoustic measurements

Although stimuli intensity had been equated at the sentence level for all talkers, the distribution of energy within sentences differed among talkers (Table 2).

	Sentence	Carrier		Target word	
Talker	Rate (syll/sec)	F_0 (Hz)	Int (dB)	F_0 (Hz)	Int (dB)
Pro	3.4 (0.2)	247 (11)	70.4 (0.5)	170 (16)	67.5 (1.3)
Worst	2.3 (0.2)	172 (8)	68.4 (0.9)	192 (35)	68.6 (1.4)
Mid	3.2 (0.3)	173 (13)	68.2 (1.3)	171 (23)	68.8 (1.2)
Best	3.0 (0.3)	190 (17)	68.7 (1.8)	187 (41)	68.4 (1.4)

Table 2: Mean values for acoustic measures of sentences (S.D.'s). Int = intensity; Pro = professional talker.

Specifically, targets spoken by the professional talker were about 1 dB less in intensity than for other talkers while the mean intensity of her carrier phrases was about 2 dB higher. Relative to the mean F_0 of the carrier phrase, the mean F_0 of target words was lowered for the professional talker but raised for the talker with the worst voice quality.

2.4 Discussion

Surprisingly, listeners performed very poorly with the professional talker, and better than expected with the talker who had the poorest voice quality. The professional talker's lower intensity on target words and the higher F_0 for the talker with the worst voice quality may have contributed to these results. Experiment 2 investigated these possibilities.

3 Experiment 2

3.1 Method

We equated the intensity of all target words and replaced the four talkers' carrier phrases with the standard NU6 carrier phrase of the talker recorded by Auditec. Listeners were 16 young adults (*M*=19.2 years, SD=2.7) who had similar characteristics as listeners in Experiment 1 and were naïve to this task. The procedure was the same as in Experiment 1.

3.2 Results

The pattern of results was similar to that of Experiment 1 (Figure 1); a within-subjects ANOVA showed a significant effect of talker, $F(3, 45) = 54.3$, $p < .001$. Correct word recognition scores for the talkers with the best and worst voices were not significantly different ($p = .17$) and their scores were higher than for the talker with the medium-quality voice (*p*'s < .01); the score for the professional talker was different from all other talkers (*p*'s < .001). A second ANOVA with 'experiment' as a between-subjects factor and 'talker' as a within-subjects factor showed that listeners performed slightly better in Experiment 1 (*M*=62.8%, *SD*=18.9%) than in Experiment 2 (*M*=58.5%, *SD*=16.8%), due to a decrease of 10 percentage points for the talker with the best voice in Experiment 2. There were significant main effects of 'Experiment', $F(1, 30) = 4.39$, $p = 0.04$, and 'talker', $F(3, 90) = 164.7$, $p < .001$, and a significant interaction between these factors, $F(3, 90)$, $p = .04$. Listeners achieved significantly higher scores for the talker with the best voice in Experiment 2 than in Experiment 1 ($p = .006$), but none of the results for other talkers differed significantly between experiments (*p*'s > .4).

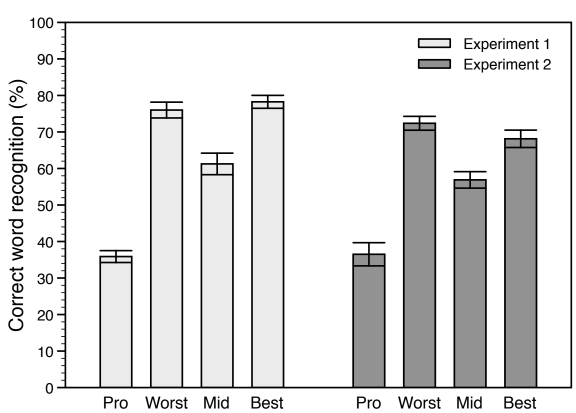

Figure 1: Mean correct word recognition scores of listeners for four talkers in two experiments.

3.3 Acoustic measurement of talkers

The talker with the poorest voice quality produced the longest duration target words of all talkers, with especially long consonant durations and long transitions between vowels and consonants (Table 3).

	Word	Consonant		Vowel steady-state portion	
Talker	Dur (ms)	Dur (ms)	Peak int (dB)	Dur (ms)	Mean int (dB)
Pro	471	63	63.1	101	72.9
Worst	717	203	68.7	98	73.9
Mid	596	143	68.7	114	72.8
Best	644	173	70.3	107	72.4

Table 3: Mean acoustic measures of target words. Dur = duration.

3.4 Discussion

Talker differences in intelligibility were not caused by differences in the intensity of target words or an emphasis on target words by F_0. The durations of transitions between vowels and consonants were longest for the talker with the poorest voice quality, followed by the talker with the best voice, with the professional talker having the shortest transitions. Since portions of the speech signal that contain change supply the most information for speech recognition [8], the differences in transition duration may have played a key role in determining talker intelligibility in this study.

4 Conclusions

Age-related changes in the voice may negatively affect speech communication, but results from this study suggest that talkers may compensate for poorer voice quality through articulation and speech rate adaptations.

Acknowledgements

This study was funded by an NSERC grant awarded to K.P.-F. (RGPIN 138472) and a CRC grant awarded to P.vL. (950-213162). We thank James Qi, Jessica Banh and Rajbir Deo for their help in experimental setup and data collection.

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AGE-FRIENDLY COMMUNITIES PRINCIPLES AND INITIATIVES

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Introduction

The present paper provides background on the global Age-Friendly Cities Initiative, as well as the Manitoban Age-Friendly Communities Initiative. The paper concludes with a discussion surrounding possible intersections between age-friendly and acoustical environments.

Age-Friendly Communities: Background

In Canada, as in many nations around the world, the proportion of older adults is increasing due to factors such as longer life expectancies and lower fertility rates. In 2012, 14.9% of the population was aged 65+, and this proportion is expected to grow such that in ten years almost 1 in 5 (18.5%) of Canadians will be aged 65+. One in four Canadians are expected to be 65 or over by 2051. It is also noteworthy that one of the fastest growing segments of the population is the aged 80+ [1-2].

In addition to the trend of population aging, globally, more people of all ages are living in cities, and this is a national phenomenon as well. In 2011, 81% of Canadians lived in urban areas as compared to 19% living in rural areas. Although variation exits across regions, in Manitoba, 72% of Manitobans lived in urban areas in 2011 [3].

The Age-Friendly Cities concept, as spearheaded by the World Health Organization (WHO), presents a response to these two major world-wide trends of global aging and urbanization [4]. To help facilitate world cities to incorporate aging issues into their city planning, the WHO initiated a global project to identify the key city features that define "age-friendly." The WHO Global Cities Project, completed between 2006-2007, involved 33 cities from across the globe, including large cities such as Rio de Janeiro, and smaller cities like Portage La Prairie, Manitoba. The project involved asking older adults and those who work with older adults to identify features they liked about their city as well as features that could be improved within the following 8 domains of city living: 1) Outdoor spaces, 2) transportation, 3) respect and social inclusion, 4) civic engagement and employment, 5) health services, 6) housing, 7) social participation, and 8) communication and information [4-5]. Results from this project were released by WHO in the Global Age-Friendly Cities Guide [4]. As well, a listing of core age-friendly features was released in the Checklist of Essential Features of Age-Friendly Cities [6]. Building on this work, Canada developed the Age-Friendly Rural/Remote Guide [7]. The research revealed similar issues as raised by the WHO project but also highlighted unique features and barriers associated with living in rural Canada.

What is Age-Friendly?

According to the WHO [4], the underlying premise of age-friendly cities is that they foster the health, security, and social participation of older adults, or what the WHO refers to as 'active aging.' The argument is that urban environments have a role to play in facilitating (or hindering) quality of life for older individuals. Specifically, in an age-friendly city, "policies, services, settings and structures support and enable people to age actively" [4, p. 5].

In addition, what has emerged from the Age-Friendly Initiative is the idea that age-friendly cities should benefit all ages, not just older individuals. For example, as reported in the WHO Guide, "According to the project participants, it should be normal in an age-friendly city for the natural and built environment to anticipate users with different capacities instead of designing for the mythical "average" (i.e. young) person. An age-friendly city emphasizes enablement rather than disablement; it is friendly for all ages and not just elder-friendly" [4, p. 72].

The checklist developed by the WHO [6] provides a concrete way of assessing the age-friendliness of cities by each of 8 domains of city living. For example, the domain of "Outdoor Spaces" pinpoints issues such as sidewalks and accessible washrooms. In the domain of "Respect and Social Inclusion" it is noted that community activities and events should allow for inter-generational participation by accomodating age-specific needs. Of course, not all of the core features are as relevant for every city: public transport would look different in larger cities than in smaller towns, for example. However, the checklist helps define what age-friendly means to older adults in each domain and can help guide city assessments and community development [5].

Age-Friendly and Manitoba

Since the initiative of Age-Friendly Cities from WHO, cities and communities around the world have begun the process of becoming more age friendly. Along with several other provinces in Canada, Manitoba has embraced the concept of creating age-friendly communities. The Provincial Government launched the Manitoba Age-Friendly Communities Initiative in 2008 which also coincided with a 5-year Community-University Research Alliance led by Dr. Menec of the University of Manitoba. Since its launch, the initiative has been fruitful, and 100 communities (including villages, towns, cities, and rural municipalities) have joined the initiative in Manitoba. Although the process has varied somewhat, for the most part, each community has formed an Age-Friendly Committee made up of community members, government, business-owners, etc. In addition, most communities have undertaken a community survey and consultation to determine the priorities for making their own community more age-friendly [8-9]. Community consultations have revealed, for example, that although there is a lot of commonalities between the needs of different communities (e.g., housing, transportation), each place also has unique strengths and unique priorities [10].

The Age-Friendly Manitoba Initiative has resulted in interesting and important linkages, for example, between city planners, government, academics, community members, and not-for-profit organizations. Given the aim of creating social and physical environments 'friendly' and accessible for all ages, it is not surprising that various stakeholders and experts need to be involved.

Discussion: Acoustics and Age-Friendly

What kind of linkages could be formed between experts on acoustics and acoustical environments and age-friendly? Specifically related to the domains of city living, it would seem that acoustics would be particularly relevant to the domains of communication, social participation, and respect and social inclusion. For example, are there elements of acoustics design that are, perhaps, hindering older adults' in participating fully in events and activities in their city or community?

Although the WHO Age-Friendly guide and checklist did not touch on acoustics per se, some examples of ways that communities were dealing with issues of hearing loss were given. One such example was taken from Portage La Prairie, Manitoba, in which older people were given the option of having a headset in church [4, p.45]. Not being able to hear properly in church represents a serious barrier to participation and involvement. On this track, the Age-Friendly New York includes in their guide for businesses a section on "sound" and what businesses can do to make the environment more comfortable for their older customers [11]. What other ways do the physical and social environment intersect with issues of hearing, communication, and social participation in older adults? And what kind of solutions could be developed?

In conclusion, the two trends of global aging and urbanization represent both challenges and opportunities. It would seem that meeting the challenges and taking advantage of opportunities will be best served through collaboration among various stakeholders.

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AUDITORY AND COGNITIVE AGING: IMPLICATIONS FOR HEARING ACCESSIBILITY

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1 Introduction

It is well known that the population is aging and that hearing loss and cognitive loss become more prevalent with age. Importantly, laboratory and epidemiological research suggests that there is a connection between hearing and memory declines in older adults. This connection is found in people who have clinically normal hearing and cognition, as well as in those who have impairments. The reasons for the connection are not yet known. Hearing problems in adverse communication environments may accelerate social withdrawal and, in turn, cognitive decline. Conversely, good communication and social interaction may be an important part of promoting healthy and active aging. This paper will provide a brief review of auditory and cognitive aging research, with an emphasis on implications for the design of accessible environments to promote and maintain active communication and social participation by older adults.

2 Auditory and Cognitive Aging

2.1 Auditory Aging

Hearing loss, defined by audiometric thresholds, is one of the three most common chronic disabilities in older adults [1]. Pure-tone threshold loss increases slowly with age [2]. By the age of 65 years, about a third have a clinically significant loss, by the age of 75 years about half are affected, and by the age of 80 most people are affected [3].

Importantly, the earliest signs of age-related hearing loss (ARHL) may be difficulty understanding speech in noise. Such difficulties due to problems in supra-threshold auditory processing problems are usually noticed before clinically significant threshold elevations are diagnosed.

There is heterogeneity in the hearing abilities of older adults. Many older adults have sensori-neural hearing loss due to damage to cochlear outer hair cells (OHCs); however, this type of hearing loss often results from exposure to noise and is not unique to aging. In ARHL, multiple structures in the cochlea and central auditory system can be damaged in ways that do not involve OHC damage and that are not typical in younger adults. The high-frequency threshold elevations typical of the audiograms of older people may result from changes in the endocochlear potentials related to damage to the stria vascularis or blood supply to the cochlea [4]. Neural damage may also occur, with associated reductions in supra-threshold auditory temporal processing, even if there is little or no change in audiometric thresholds [5]. Thus, there is much variability in the degree of difficulty that older listeners experience in everyday life. Their speech-in-noise performance is often poorly predicted from the audiogram and their problems may not be solved readily by simply amplifying sounds.

2.2 Cognitive Aging

As healthy older adults age, some aspects of cognition get worse, but others get better [6]. There are gradual declines in the ability to process information such that it becomes more difficult for older listeners to attend to, understand, and remember what they have heard. However, there are also gradual gains over the adult lifespan in knowledge of the world, linguistic knowledge (e.g., vocabulary), and other types of expertise. Older listeners may compensate for difficulties processing information by using knowledge to advantage. Beyond normal age-related cognitive changes, clinically significant mild cognitive impairment and dementia increase with age, such that the age of 70 years, about a fifth of people have a clinically significant cognitive loss [7].

2.3 Auditory-Cognitive Links

Adults of any age have more difficulty paying attention and remembering when they listen in noise compared to when they listen in quiet. Accordingly, at least some of the cognitive problems of older adults are aggravated by ARHL and the mental challenges of listening in noisy situations [8]. Over decades of living with hearing loss, ARHL also seems to put people at greater risk for dementia [9-11]. Indeed, some studies have found that individuals with hearing loss had a 2 to 5 times increased risk of developing dementia [12] and that for every 10 dB of hearing loss over 25 dB HL, individuals had a 20% increase in their risk of developing dementia [13]. More research is needed to discover the reasons for the links between hearing loss and cognitive loss [14].

Some evidence suggests that hearing rehabilitation could contribute to cognitive health [15], but well-controlled studies to test this idea are just beginning. In the meantime, seeking help for hearing problems may be a good idea. It is also more important than ever to consider the special auditory and cognitive needs of older adults when designing hearing accessible acoustical spaces for communication consistent with age-friendly social policies.

3 Listening Needs

3.1 Auditory Considerations

To match the performance of younger adults on tests of word recognition in multi-talker babble, older adults with good audiograms typically need a signal-to-noise ratio (SNR) that is about 3-4 dB more favourable, and those with high-frequency hearing loss typically need an SNR that is about 9 dB more [16]. Of course, energetic masking occurs when the spectra of the target speech and the masker are similar. However, the temporal properties of the masker are also important to consider because older adults often have reduced auditory temporal processing abilities. Compared to listening situations in which the background noise is relatively steady, age-related differences are greater if the background noise fluctuates because it is more difficult for older listeners to detect temporal gaps and to glimpse a target speech signal in the gaps of a fluctuating masker. Furthermore, age-related differences are even greater if the background noise is the speech of one talker, partly because the content of what is being said by the competing talker may be distracting, and partly because it becomes more difficult to segregate speech streams when their spectral and temporal acoustical properties are more similar, thereby increasing 'informational' masking. In particular, age-related difficulty increases if the target talker and competing talker(s) are of the same gender, probably because older listeners can have reduced ability to use periodicity cues to separate speech streams based on between-talker differences in the fundamental frequency and harmonic structure of the voices [17]. In terms of using binaural cues, older adults are also less able than younger adults to use spatial separation between a target and competing talker to separate speech streams [17].

3.2 Cognitive Considerations

Problems hearing in challenging conditions are compounded because the listener must allocate more cognitive resources to pay attention, understand, and remember information that has been heard. The demands on attention are even greater when listening is not the only task. Even ordinary multi-tasking such as listening while walking (possibly when using a mobility device such as a walker) can require the person to divide attention to an extent that may be related to the finding that older people with hearing loss are at greater risk of falls compared to peers with good hearing [18].

4 Conclusions

In general, hearing accessibility improves if communication environments are less noisy and reverberant. For older adults, in addition to reducing relatively steady background noise sources, it will be important to design environments to facilitate the ability of listeners to focus on a target talker and ignore the distracting, competing voices of others sharing the same space. Acoustical designs may also need to take the multi-tasking demands on older users into account if they need to move or do other tasks while listening.

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RECENT TRENDS IN THE ACOUSTIC DESIGN OF INSTITUTIONAL FACILITIES

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1 Introduction

Acoustics continues to be an important factor in the design of institutions such as Healthcare Facilities, Schools and Retirement Complexes, through the technical innovation and the implementation of Design Standards such as LEED for New Construction and The Green Guide for Health Care (published by the U.S. Green Building Council), and the Sound and Vibration Design Guidelines for Hospital and Healthcare Facilities (published by the Facility Guidelines Institute and adopted by the American Institute of Architects and Academy of Architecture for Health). This article discusses new innovations and directions in the field in the context of the authors' professional practice.

2 Learning Institutions

ANSI S12.60-2010 American National Standard, "Acoustical Performance Criteria, Design Requirements, and Guidelines for Schools, Parts 1 and 2" [1], provides guidelines and acoustical criteria for the design of classrooms and other learning spaces. While not yet endorsed by the CSA for use in Canada, it is an excellent reference. For learning spaces it is very important that the interior acoustics of the room be well controlled so that speech is clearly audible. This is particularly important for young children and persons with hearing, language, speech, attention deficit or learning disabilities. The document provides criteria for acceptable levels of reverberation, background sound from mechanical systems, and noise isolation from both indoor and outdoor spaces. In addition it provides helpful guidelines and design examples to assist school planners in meeting those criteria.

Schools are built under considerable budget control and the standard provides a benchmark for defining realistic requirements, conducting value engineering and avoiding cost overruns. It helps clients understand which elements of a design are most important, avoiding the tendency to either "overdesign" elements or cut critical features.

3 LEED Designs

Leadership in Energy and Environmental Design (LEED) is a set of rating systems for the design, construction, operation, and maintenance of green buildings. Many institutional facilities are designed to achieve LEED Certification, but simply scoring LEED points without considering the acoustical implications can result in some fairly significant acoustical challenges [2].

3.1 Light Penetration

Light wells, atria, openings in partitions and glazing are used to allow natural light to penetrate deep within a building. These features also allow the propagation of sound from place to place, creating problems for those who need high privacy or confidentiality. Large glazing areas and hard surfaces can result in excessive reverberation, occupant noise poor privacy and distraction if not addressed through the creative use of sound absorbing acoustical treatments.

3.2 Energy-Efficient HVAC Systems

The current trend toward using radiant surfaces for heating and cooling means that HVAC systems are significantly quieter than in the past with the level of background noise reduced to below sound masking levels. The use of alternative, energy-efficient HVAC systems may also result in poor privacy, particularly if ceilings are hard surfaced and floors are left unfinished or tiled.

3.3 Sound Masking Systems

Are libraries quiet? It's a question of terminology. When we say that a library should be quiet, we are really saying that patrons should enjoy freedom from distraction. They don't want to hear every page turn. Similarly in an office, most people don't want to hear what everyone is saying. People learn to adopt an "office polite" speech level and use headsets instead of speakerphones.

As a result, sound masking systems are becoming an integral part of office design. In fact, recent research conducted by the National Research Council suggests background sound such as provided by a well designed sound masking system to be one of the important elements of open plan offices [3],[4]. At this point in the ongoing development of LEED, sound masking systems are not credited with a point score except in heath care facilities.

3.4 Regional, Renewable and Recycled Materials

The ongoing trend toward lightweight building materials can result in less sound reduction and more noise intrusions. This applies to the building envelope as well as interior partitions. Outdoor traffic noise can cause audible intrusions, and the transfer of noise between spaces. Bass sounds are the hardest to control, as there is no real substitute for mass. The sound transmission class (STC) rating of a masonry and a drywall construction may be the same, but the low frequency sound transmission through drywall is greater, a factor building codes do not address. Similarly, impact isolation (footfall or impact noise) is not directly addressed in building codes or LEED guidelines.

3.5 Over-Reliance on Electronic Sound Systems

Most auditoria, lecture theatres, or other spaces where speech must be audible at a distance are provided with sound systems for A/V and voice lift purposes. It is important to be careful that good natural acoustics are not overlooked because of an over-reliance on electronic supports. Sound systems should be designed to complement existing room acoustics rather than fix acoustical problems. In general, sound systems are designed for people with normal hearing and do not compensate for hearing loss.

Of course, the proper use of electronic system is very appropriate in many settings. For example, building codes generally require Hearing Assistive Listening systems for public spaces with large occupancies. Electronic paging and call systems are also very useful in the design of Health Care Facilities as discussed below.

4 Health Care Facilities

In recent years, a joint working committee of the ASA, INCE and NCAC has created a set of acoustical standards for Health Care settings, included in a set of Sound and Vibration Design Guidelines which are published by the FGI. These guidelines were incorporated into the FGI Guidelines for the Design and Construction of Health Care Facilities [5] in 2010.

These guidelines provide a standard basis from which sound- and vibration-related construction specifications for health care facilities can be developed. They are also referenced in the USGBC's Green Guide for Health Care [6], where it is suggested that up to two complementary construction credits (equivalent to LEED credits) can be obtained by implementing some of these standards. However, meeting these requirements can be onerous, and in the authors' experience, design teams generally do not pursue such credits as they are not considered to be worthwhile. Further, there are several significant difficulties to consider in implementing practical designs to achieve these targets, many of which are only touched on or not discussed at all in the Design Guidelines. Some examples are given in the following sections.

4.1 Fibre Free Ductwork

Hospitals and many other health-care facilities do not allow fibrous materials inside ventilation ductwork. Typical engineered duct liner materials, which are treated with a mat facing to prevent the migration of any fibrous materials into the airstream, are often considered inadequate to ensure that organic materials cannot get trapped and start to decay or grow within the ductwork. This restriction on the use of products commonly employed in other types of construction makes it difficult to achieve the low background sound level targets outlined in the Guidelines. Alternative types of acoustic attenuators, such as packless silencers and foam rubber duct liners, are often integrated into the design, but these tend to be less effective and more costly, and there are some practical limitations to what can ultimately be achieved.

The same restrictions can also lead to compromises in the effective sound insulation ratings that can be achieved between adjacent rooms due to sound flanking through ductwork. In many other applications, acoustical duct liner in common branch ducts or run-outs is often added to prevent such effects. Alternative methods that must be considered for health care settings include longer run-outs between branch ducts and terminal boxes with multiple elbows, which generally requires more space and materials, and non-metallic flexible ductwork, which can only be used in low-pressure systems.

4.2 Security Concerns

In some health care settings, particularly those related to mental health, security concerns are paramount. Walls that would normally be constructed with light-gauge metal studs are built with concrete block, heavy gauge structural studs and/or plywood sheathing, to prevent damage. Spaces that would normally include accessible t-bar ceilings with acoustic tiles are designed with drywall or cement board instead. This can lead to walls that have reduced STC ratings unless special acoustic designs are implemented, and rooms that are acoustically 'hard', exhibiting poor interior acoustics. Surface acoustic treatments (such as acoustic panels) sometimes cannot be used for similar reasons. In some cases, due to these restrictions, the recommended targets outlined in the Guidelines cannot practically be met.

4.3 Paging and Call Systems

The Guidelines describe acoustical design targets for overhead paging and call systems to ensure good speech intelligibility. To further enhance the acoustic environment, the Guidelines recommend that consideration be given to replacing overhead paging, locating and call systems with new wireless technologies, such as communication badges, vibrating beepers, RFID and infrared. Most new hospitals are in fact integrating such technologies, and reducing reliance on overhead systems.

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Dr. Leo L. Beranek
Westwood, MA
USA

September 15, 2014

Dear Leo –

You truly are an inspiration! Your work has brought so much insight to the world of acoustics and concert halls. And at 100 years, you are clearly going strong. May you have many more years of happiness, good concerts, and successes ahead!

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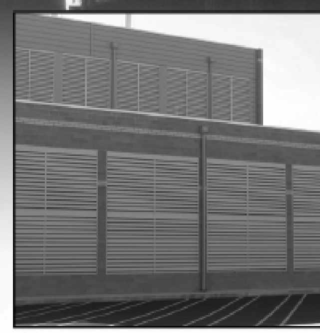

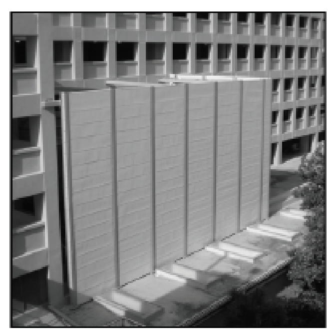

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